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Systems Control, Inc.

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IMPLICATIONS OF DEMAND ASSIGNMENT FOR FUTURE SATELLITE COMMUNICATION SYSTEMS

FINAL REPORT

PREPARED FOR

DEFENSE COMMUNICATIONS AGENCY
DEFENSE COMMUNICATIONS ENGINEERING CENTER

1860 WIEHLE AVENUE
RESTON, VIRGINIA 22090

UNDER CONTRACT
DCA100-76-C-0060

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June 1977

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Defense Communications Agency
Defense Communications Engineering Center
1860 Wiehle Avenue
Reston, Virginia 22090

Under Contract:

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The objectives of this study are (1) to investigate the system-wide implications of using Demand-Assignment Multiple-Access (DAMA) techniques with communication satellites to serve a diverse community of voice and data communication users, and (2) to explore extensions of the current DAMA techniques to Random Multiple Access (RMA) techniques which are suitable for data and packetized voice. The major emphasis of the study is exploring system-wide cost-performance tradeoffs in order to determine:

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- The most promising DAMA/RMA techniques,
- The magnitude of potential cost savings,
- The optimal satellite/terrestrial mix, and
- Areas requiring more definitive study.

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1 INTRODUCTION

1.1 GENERAL

This study is an investigation of the system-wide implications of using Demand-Assignment Multiple-Access (DAMA) techniques on communication satellite systems to serve a diverse community of voice and data communication users and to explore the suitability of extensions of DAMA technology to Random Multiple Access (RMA) techniques for use in future integrated voice and data DCS switched, common-user networks.

The investigation was conducted under DCA Contract DCA100-76-C-0060 as an integrated team effort, led by Systems Control, Inc. (SCI) with support from the Stanford Research Institute (SRI) and the Ford Aerospace and Communications Corporation (FACC). The study was a one-year effort at about a two man-year level-of-effort.

1.2 SCOPE OF REPORT

This report is the final report covering all of the study and incorporates all the results previously published in draft form which covered Task I (Categorization of Demand Assignment techniques) and Task II (RMA Techniques Applied to Voice Users).

1.3 STUDY OBJECTIVES

The global objectives of this study are (1) to investigate the system-wide implications of using Demand-Assignment Multiple-Access (DAMA) techniques on communication satellites to serve a diverse community of communication users, and (2) to explore the suitability of extensions of the current DAMA technology to Random Multiple-Access (RMA) techniques. A major emphasis of the study is on exploring the system-wide cost/performance tradeoffs of alternative DAMA-satellite techniques in order to understand:

- The most promising DAMA techniques
- The magnitude of potential cost savings
- The optimal satellite/terrestrial mix
- Areas requiring more definitive study.

1.4 SCOPE OF STUDY

The scope of this study is limited to analytical investigations and computer evaluation of complex analytical models of DAMA techniques, and to creative extensions of current DAMA approaches, particularly in the area of non-orthogonal RMA modes of access. Since previous packet-switching RMA concepts have been motivated by the requirements of data communication, there is almost no published work in applying these concepts to voice communication. This study investigates the utility of applying RMA techniques to voice communications.

Specifically, the study contract was divided into four tasks with scopes as follows.

Task I: Categorization and characterization of demand assignment techniques, applicable to the common-user DoD military communications environment, particularly in terms of the total communications system implications on large-scale common-user switched channels requiring satellite paths and many ground stations.

Task II: Investigation of non-time/frequency orthogonal modes of multiple access demand assignment, particularly the extension of pure random access techniques (of which the "ALOHA" technique is a prime example) to voice system multiple access.

Task III: Investigation of channel utilization improvements attainable through source encoding of digital speech on a random access voice multiple access channel.

Task IV: Suggestion of additional investigations or study areas which have promise of providing for the ability of future satellite systems to meet large-scale common-user communication system requirements in a cost effective way, and to develop evolutionary approaches to meeting future common-user requirements with satellite system implementations.

1.5 TASK/REPORT RELATIONSHIP

Since this report is organized into chapters which are felt to be a more natural presentation of results than the contract task structure Table 1.1 is included as a cross-reference between the four tasks and relevant chapters of this report.

TABLE 1.1 TASK/CHAPTER CROSS-REFERENCE

	Chapter								
Task	2	3	4	5	6	7	8	9	10
I	•	•	•	•	•		•		•
II	•		•	•		•	•		•
III	•								•
IV						•		•	•

1.6 SUMMARY OF METHOD OF APPROACH

Figure 1.1 summarizes the approach of this study by identifying the major issues for each of the four tasks. The Task I issues are broken down in additional detail to illustrate the interrelationships.

As indicated in the figure, Task I establishes the overall study framework. While Task II is the nominal beginning of analysis of random multiple access schemes applied to voice, it was necessary to move ahead of schedule in this area in order to establish this structure during the Task I effort. Additional work to extend RMA techniques for voice, establish suitable performance measures and system performance/cost tradeoffs is then incorporated into the framework established in Task I.

As originally formulated, the objective of Task III was to establish what improvements in the channel untilization/performance tradeoffs could be achieved for the packetized voice RMA systems. As the study progressed it became clear that more information on human communication performance measures for RMA systems must be experimentally established before such trades are realistic. Accordingly the emphasis in Task III shifted to a pilot experiment to bracket the performance measures.

Task IV develops an evolutionary approach to system implementation based on the system-wide cost tradeoffs developed in the Task I framework. It also organizes the open questions which have been generated in this study into the form of specifications for several studies/experiments that indicate promise in resolving these questions.

1.7 SUMMARY OF THE REPORT BY CHAPTER

It is evident from Figure 1.1 that the amount of interaction and parallism among topic areas of this study makes it difficult to choose

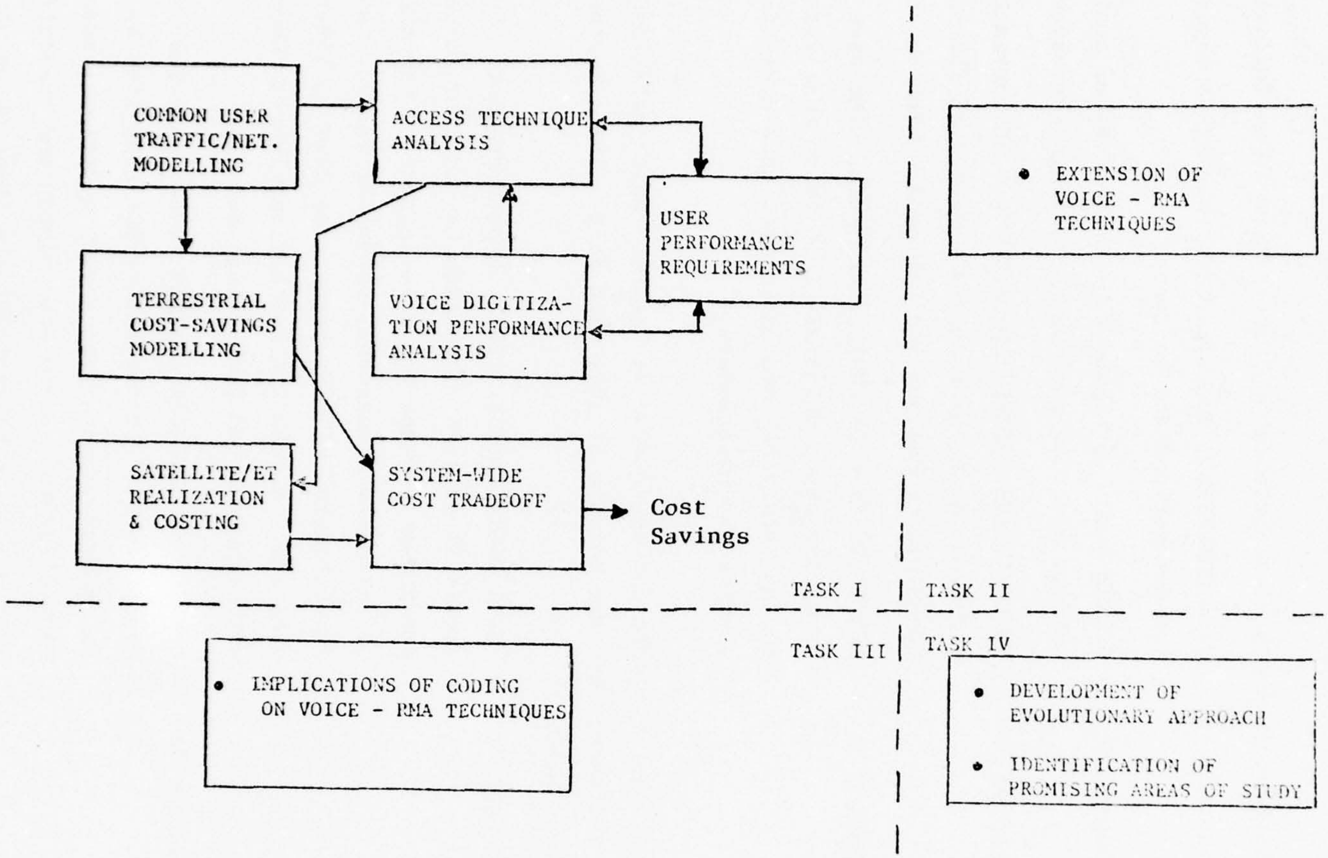


FIGURE 1.1 SUMMARY OF STUDY APPROACH

a "best" sequence of chapters to present the material. However, to aid the reader, a brief guide to the report by chapter is presented below.

The most direct logical path through the report is as follows:

Chapter 4: DAMA CONCEPTS. The alternative DAMA/RMA concepts are categorized and analyzed.

Chapter 6: SYSTEM-WIDE COST TRADEOFFS. The level of cost savings and best deployment strategy are established.

Chapter 7: EVOLUTIONARY ISSUES AND APPROACH. An approach is proposed which incorporates elements of the best candidate techniques and can evolve with the DCS.

Chapter 9: IDENTIFICATION OF FURTHER RESEARCH. The open issues from this study which are felt to be most important to the future DCS are summarized into suggested studies and experiments.

The other chapters can be treated as "asides" which establish background and expand certain details of this main line. Their contents is summarized below.

Chapter 2: VOICE DIGITIZATION. This chapter presents the background in voice digitization or encoding issues and perofrmance needed to fully appreciate the choices of performance measures, parameter values, and "suitability" of DAMA techniques used in later chapters. APPENDIX A further supplements Chapter 2 with experimented methodology details.

Chapter 3: COMMON USER NETWORK TRAFFIC MODEL. A common user traffic model for the future DCS (supplied by DCA) is presented and expanded with additional assumptions needed for the system comparisons and tradeoffs in this study. A terrestrial transmission network cost model is developed for use in the system-wide cost tradeoffs.

Chapter 5: COMMUNICATION SATELLITE REALIZATION ISSUES. The earth terminal/satellite system design realization issues, relationships, technology risks, and costing for the alternative DAMA concepts is presented. This chapter is also supplimented by APPENDIX B, which documents details of the costing methodology.

Chapter 8: CONTROL STRUCTURE ISSUES. A systematic summary and comparison is given of the control issues among alternative DAMA concepts.

1.8 CONCLUSIONS

1.8.1 System-Wide Implications

For the large common-user predominately voice traffic future DCS (1980-1990) baseline traffic/network model, we conclude that:

1. A single large DAMA satellite system is capable of providing all inter terminal area connectivity for the CONUS network. A number of DAMA techniques are feasible - the primary choice depending on whether the digitized voice is packetized or bit stream.

2. Such a system can produce annual cost savings in the $70-100 million range.

3. O&M costs for earth terminals dominate all other system costs and determine the best deployment strategy and level of savings. This conclusion should mandate a new emphasis on automation of earth terminals, as well as a new look at possible savings of automating other key DCS elements.

1.8.2 Dama Technique Categorization/Comparisons

In comparing the broad range of circuit-switched and packet-switched DAMA techniques, we conclude that:

1. Packet-switched techniques begin with an originated traffic advantage over circuit or bit stream techniques of about 2:1 due to voice duty factor.

2. Dispite the inherent voice duty factor advantage of packetized voice, the ALOHA-based techniques with delay and packet loss suitable for voice are not efficient enough to be preferable to circuit-switched techniques.

3. A new packet-switched technique called destination variable access (DVRA), originated in this study, is the most bit rate efficient technique investigated. The new technique fully utilizes the 2:1 voice duty factor advantage over any circuit-switched technique and should be further developed to the point of feasibility demonstrations.

4. The original retransmission(packet conserving) ALOHA protocol is not suitable for voice over satellites and is inherently poor in stability, securability and vulnerability.

1.8.3 Voice Digitization/Encoding

The effects of different choices of voice digitization or encoding techniques and performance measures are fundamental to the comparison and design of future satellite voice communication systems - especially when packet-switched. The pilot experiment in this study has provided the following guidance:

1. For systems with packet losses, packets should represent short (20-50 ms) speech times.

2. Packet retransmissions over satellite links should be limited to about one retransmission.

3. Forward error correcting codes are important to preserving packet header integrity, but not important to speech at bit error rates normally designed into satellite links which support data.

4. Expanded communication task experimentation is needed. Such experimentation will be most profitable if conducted in connection with a communication system study.

1.8.4 Technology Risks Issues

Investigation of the feasibility and technology risk associated with the various DAMA techniques resulted in the conclusions that

1. The preferred techniques (fully variable and destination variable) can utilize existing access technology (e.g., SCPC and TDMA).

2. Both large bandwidth and cost effectiveness push toward Ku band in future large systems.

1.8.5 New Techniques Originated In Study

In the course of the proposal effort and in the study itself a number of new DAMA techniques and system approaches were originated. These contributions are summarized below.

1.8.5.1 New ALOHA-Based Protocol Concepts

In response to the inherent shortcomings of the original, computer-data based ALOHA concept for satellite voice two-way communication, a number of new ideas were proposed and analytically investigated. These variations modified the original ALOHA protocol to be more suitable for our application, and are:

- No-retransmission ALOHA
- Finite-retransmission ALOHA
- Multiple copy ALOHA

1.8.5.2 New ALOHA-Based Implementation Concepts

In addition to the modifications of the ALOHA protocol to meet the needs of voice, two new implementations techniques, which can apply to any of the above, as well as the original ALOHA protocol were originated. These were:

- Spread spectrum ALOHA
- Capture processing satellite for ALOHA protocols.

The capture satellite stabilizes the original protocol and improves efficiency for all modifications.

1.8.5.3 Destination Variable Random Access

The most promising and efficient technique investigated, called destination variable random access (DVRA), was originated in this study. In this technique, a temporally fixed fraction of total satellite capacity in a TDMA burst is allocated to each earth terminal. Packets are buffered at each terminal and re-broadcast by the satellite in the allocated TDMA burst. Each earth terminal "listens" to all bursts and detects those packets destined for it by reading the header information.

1.8.5.4 Destination Variable Packet/Circuit Hybrid

A hybrid combination of packet-switched DVRA (above) and circuit-switched destination variable demand access (DVDA) was devised and proposed as an approach to building a system for the evolving DCS, which efficiently handle <u>both</u> packet traffic and traffic that must be circuit-switched to maintain bit stream synchronization.

2 VOICE DIGITIZATION

2.1 SURVEY OF VOICE DIGITIZATION TECHNIQUES AND ISSUES

Methods of digitizing voice signals have been studied extensively. A considerable number of such technqieus have been suggested. In this section we briefly discuss the most important generic voice digitization techniques and their suitability for voice transmission in a communication system such as the DCS.

In general, any method of digitizing voice produces an inexact representation. There is a trade-off between the fidelity of the representation and the number of bits used to represent a given duration of speech. Various methods differ in the type of degradation that they introduce. Thus, user requirements must be carefully examined in order to choose between methods.

In a communication system there are other practical considerations that help one to choose among methods. The hardware costs can be of great importance. Since bit errors are bound to occur, their effect on the quality of reproduced speech can be critical. This is especially true when a satellite node is a part of the transmission path, since for ALOHA satellite systems, retransmission delays are highly undesirable.

There are two main classes of digital methods for voice representation. The first such class entails direct encoding of the speech waveform or a function of the waveform. Pulse code modulation (PCM) and its variants fall into this category. The second class consists of an encoding of characteristic features of speech that result from the physiology of the human vocal system. These are the so-called VOCODER methods.

2.1.1 Direct Encoding Methods

Normally, it is sufficient to transmit the portion of voice that falls between 0 and 4 kHz. The signal is usually sampled 8000 times per second and the samples are quantized. The transmission bit rate is thus 8000 times the number of bits used to represent each sample. The reproduced signal is an imperfect representation of the actual signal because of the noise introduced by the quantization. The more bits per sample, the better the quality of speech reproduction.

The voice signal can be encoded by a variety of means, including:

- PCM, in which quantized samples of the voice signal are transmitted.
- DPCM (differential PCM), in which a quantization of the difference between the previous and current value is sent.
- Adaptive quantization schemes, in which the quantization levels are changed dynamically to achieve entropic encoding matched to the particular speech sample.

All these methods incur quantization noise because each sample is encoded with a finite number of bits.

A simplification of the adaptive quantization schemes is Continuously Variable Slope Delta Modulation (CVSD). In this method, each sample is encoded with 1 bit. The quantization levels are continuously adapted according to the digital bit stream: Three successive ones increase the level, and three successive zeros decrease the level. The decoder uses the same algorithm. Sample rates two to three times the Nyquist rate of input lowpass filter generally give good quality speech. The widespread use of CVSD is due to the simplicity of the algorithm; the ease of varying the bit rate (by changing the sample rate); and the preservation

of speech quality if bit errors, background noise, and lost data occur. Since the bulk of wideband testing has been done on CVSD-encoded speech, we focus on CVSD in this report.

2.1.2 Vocoder Digitization and Bit-Coding Structure

Vocoders substantially reduce data rate by modeling the production of spoken sounds as a time-varying filter excited by uniform spectrum signals. The time-varying filter accounts for all frequency shaping corresponding to glottal waveforms, vocal tract configuration, and radiation effects. The usual method is to assume that the spectral parameters are constant over a short period of time called a FRAME, and then to estimate them by using standard frequency and or correlation analysis techniques. Generally, 10 to 14 parameters describe speech short-term spectra for frame sizes of 15 to 25 ms. The coding of the excitation signal distinguishes two classes of vocoders: pitch-excited (PEV) and voice-excited (VEV). The PEVs give low rates by representing the glottal excitation as pulses during voiced sounds and as noise during voiceless sounds. Two examples of PEVs are the Belgard channel vocoder and the linear-predictive coefficient (LPC) vocoder. These two PEVs differ in the manner by which the spectral information is extracted and represented. Both are under investigation to find low-cost implementations with LSI circuitry. A VEV uses the spectral filter to derive a reduced bandwidth signal and then uses a PCM method to code this signal. The Residual Encoded Linear Predictive (RELP) Vocoder is an example of this class and gives very good quality speech at data rates of 6 to 10 kbits/s.[47]

The estimated parameters for each frame are coded into a PARCEL with excitation parameters (the PCM code for the particular frame for VEVs, or pitch and gain for PEVs) and the spectral parameters. Each parameter is assigned a number of bits, depending on its relative contribution

to overall speech quality. A typical frame allocation for an LPC PEV is 5 bits for gain, 6 bits for pitch, and 10 spectral parameters, with 7, 7, 6, 6, 5 . . . 5 bits -- giving a total of 67 bits per frame. For a 50-frame/s rate, the data rate is 3.3 kbits/s.

2.1.3 Comparison of Speech Digitization Techniques

There are many variations of the PCM and vocoder digitization techniques, and a comprehensive overview is not in order here. For the purposes of this study, generic rather than specific differences are relevant. Table 2.1 lists some important differences between narrowband (PEV) and wideband (CVSD) voice digitization techniques.

TABLE 2.1 VOCODER PERFORMANCE PARAMETERS

Parameters	PEV	CVSD
Cost	\$200 - \$8000	\$5 - \$100
Bit Rate	2K - 5K	10K - 16K
Intelligibility Quiet Room	86% *	82%
Intelligibility Noisy Room	55%	74%
Bit Error Rate Curve Knee	0.1%	1.0%
Packet Loss Degradation	1% (high quality) 10% (useability)	1% 10%
Lost Speech Recovery	Good	Poor

*All intelligibility scores are MRTs.

The two most significant differences are cost and bit rate. Projection of digital-signal-processing and LSI technologies indicate that the CVSD algorithm can be implemented in a few chips. Mass production can reduce the price of a complete voice digitizer to less than $100. The current PEV prototype signal processors are standard programmable CPU architectures with very high speed memories and ALUs. The memory costs will continue to dominate the total cost for the next ten years, so prices of $2000 to $8000 are expected for that time frame. DARPA has recently begun a new program to use the CCD memory technology for development of low-cost vocoders. The goal of that program is to produce a small vocoder terminal that has equivalent intelligibility of the phone system and costs less than $1000. If the quality of current vocoders can be preserved (or improved) with these low-cost vocoder-terminals. The cost difference between PEVs and CVSD will be significantly reduced. The ranges of bit rate are 2 to 5 kbits/s for PEV and 10 to 32 kbits/s for CVSD. The speech quality is barely acceptable at the lower ends of these rate ranges.

The voice quality of the two techniques is comparable for very good background noise environments, but the PEV quality rapidly deteriorates when the input SNR decreases. This is one of the more serious limitations of current PEV techniques. The nominal quality is not degraded further by channel bit errors with rates less than 0.1% for PEV or 1% for CVSD (the nominal - perfect channel - quality is determined by the vocoder type, bit rate, background noise and audio signal conditioning). Another way in which transmission channel performance can affect speech quality is failure of segments of the speech bit stream to be received, for instance, when packets are lost in an ALOHA system. The point at which speech degradation becomes noticeable is a function of many parameters, but it is in the range of 1% to 5% lost packets.

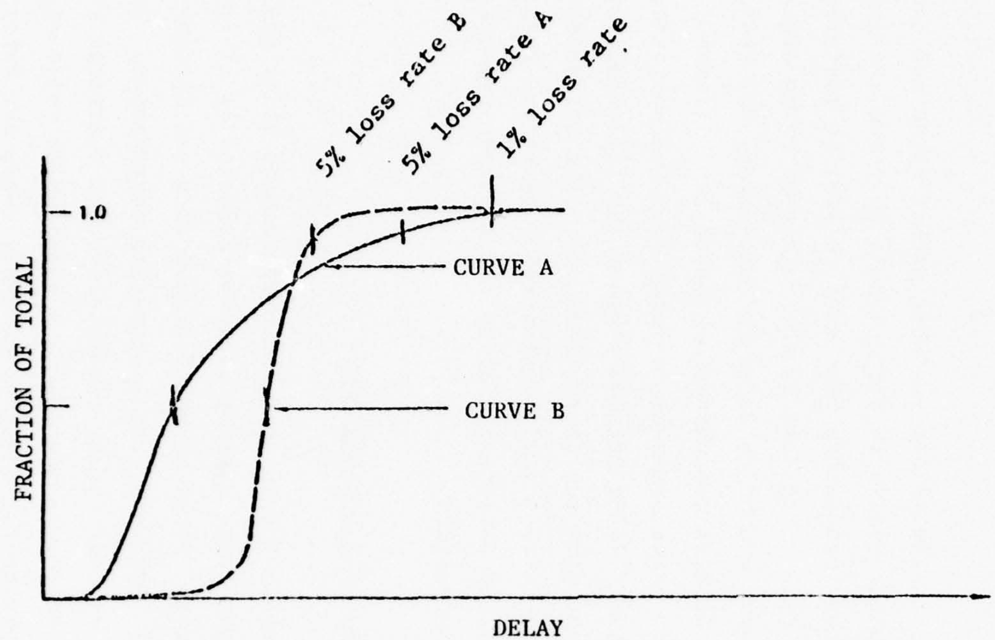
Another point of comparison is recovery of lost speech. When gaps occur in the reconstructed speech, due to lost or blocked packets,

the resulting choppiness can be very objectionable. A commonly proposed method for improving voice quality in these situations is to smooth or interpolate the speech through these gaps. If the duration of the gaps is sufficiently short (on the order of 10 to 30ms.), PEVs, such as LPC vocoders which are based on short-term stationary speech models, allow such smoothing. The PCM methods do not permit interpolation as easily because of their sample-by-sample encoding.

Table 2.1 indicates a large gap in bit rate between 5 and 10 kbit/s. For this range, several speech digitization methods have been proposed, but extensive testing has not been done. The complexity and modeling characteristics of one such system, RELP, are much the same as those of PEVs, with robustness and intelligibility as good as or better than those qualities of CVSD.

2.1.4 ARPAnet Speech Packet Transmission Experiments

SRI has participated in several packetized speech experiments using the ARPAnet. Since the ARPAnet was designed primarily for data transmission between computers, the design imposed certain constraints on the voice transmission [20]. The subnet (IMP-to-IMP) required error-free transmission of packets. When errors on the subnet were detected, hop-by-hop retransmission occurred which resulted in variable delays. During the speech experiments, it was found that end-to-end acknowledgments introduced very long delays and hence they were not used. Thus the voice transmission system is error-free with packets being lost when their transmission delay exceeds a fixed amount. The important system characteristic is the probability of packet delay exceeding a given threshold, and not average delay, as can be seen in Figure 2.1. The common strategy is to set a fixed reconstruction delay at the receiver, so that, say, 95% of the packets arrive in time. As Figure 2.1 shows, systems with different delay distributions have different reconstruction delays and hence differ in performance.



(A) SMALL AVERAGE DELAY BUT LARGE DISPERSION
(B) LARGE AVERAGE DELAY BUT SMALL DISPERSION

FIGURE 2.1 CUMULATIVE DISTRIBUTION FUNCTIONS FOR TWO DIFFERENT CASES

An aspect of packet voice that is currently being studied is the effect of temporal distortion on intelligibility and speech quality. There have been numerous studies on the effect of channel errors on vocoder intelligibility, but since the ARPAnet gives no channel errors and introduces gaps only because of delayed or lost packets, these studies have concentrated on the temporal effects. The results are discussed in Section 2.3.

Packet voice transmission allows variable packet sizes and hence asynchronous data rates not available with circuit switched transmission. In the ARPA studies, this feature has been used to acheive additional data rate reductions. Each parcel can be compressed by using algebraic techniques such as Huffman coding to increase the entropy for the coded bits [15]. Also, the inherent redundancy in speech can be removed by determining whether each parcel must be transmitted. One variation of this approach, called DELCO [47], decides whether the following parcel is sufficiently different from the transmitted one. The criterion used to determine the difference is a quality-preserving measure. Data rate can be reduced 30%, without affecting quality while larger reductions entail some sacrifice in quality.

2.1.5 Types of Voice Quality Degradation

In determining overall system cost and performance, it is most important to consider how voice quality would be affected by the various candidate DAMA techniques. Three types of degradation can decrease voice quality: temporal, channel, and compression degradation. It is largely true that temporal and channel degradation are functions of DAMA technique, while compression degradation is a function of voice digitization technique and environmental factors; however, there are interactions among the three kinds of degradation.

Temporal degradation is exhibited by such phenomena as the loss of some segments of the speech stream, delay of speech from talker to listener, and interruptions of various length in the speech stream. Although extensive studies have not been performed, it is known that, for otherwise high quality speech, some degree of temporal degradation is quite tolerable. There is evidence [12] that in two-way conversations, fixed one-way delays of up to 300 ms are tolerable and are probably not even noticeable. Similarly, approximately 1.0% of 20-ms segments can be lost with virtually no effect on quality [21]. Regularly spaced interruptions of 200 ms every 500 ms while obviously making the speech sound somewhat unnatural, also apparently do not affect intelligibility [29]. Although it is known that such minimal degrees of degradation do not significantly affect conversational speech quality, extensive parameterization studies of degree of degradation as a function of these factors and their interactions have not yet been done. Voice quality is known to be a function of the segment loss rate and the segment time, that is, the amount of speech playback time each segment represents. For the DAMA systems under study here, the segments of speech correspond to a packet, and that terminology is used.

Figure 2.2 shows a hypothetical relationship among some quantitative measures of voice quality, packet time and loss rate. The dimensions for these measures are not known, but some general features should exist. For very small packet times, the relationship between quality and the packet loss rate (Pl) should look like that between quality and channel bit error rate (Pe) as packet length approaches 1 bit. For no packet loss (infinite retransmission ALOHA system), quality is independent of packet time (the overall system delay is still important, since it affects user acceptability). There should be a plateau for small packet times such that for a given Pl there is a range of packet times (0 to x ms) that have approximately the same voice quality. Past that plateau, the curves should decrease monotonically. This idealized view will not necessarily be the case. If the quality measure were some kind of

preference test such as PARM, users might prefer a system with very large packet times. In this instance, with low P1, a listener would hear long segments of nominal quality speech with occasional deletions of whole words. This study considers such cases as different categories of service to be selected by the user and concentrates the trade-off study on the portion of Figure 2.2 that is shown.

Channel degradation is obviously a function of the network DAMA technique. For example, an infinite-retransmission pure ALOHA packet network would be expected to have zero channel errors, while a one-transmission network with no error checking or correcting, operating under marginal conditions, might have a very high bit error rate. The bit error rate is a function of the transmission network, but the effect of the errors upon speech is a function of the voice digitization technique being used in the network. For relatively straightforward wideband encoding techniques such as 16 and 32-kbits/s CVSD methods, rather large bit error rates may be tolerable. Intelligibility is preserved up to error rates of at least 10%, and quality seems to be very little affected up to error rates of 1%. On the other hand, more sophisticated narrowband systems such as 2.4 kbits/s LPC vocoders, begin to suffer a significant decrease in intelligibility at bit error rates greater than 1% and decreases in quality at error rates greater than 0.1%. Thus it seems reasonable to attempt to specify networks that will not be expected to have channel error rates in excess of 1% for wideband transmission links or 0.1% for narrowband transmission links.

Compression degradation occurs because of the limited bandwidth in transmission networks. To digitize and transmit speech with no noticeable decrease in quality or naturalness from ordinary telephone speech requires about 50 kbits/s. Current understanding of speech production and perception processes is such that the speech signal can be modeled so as to produce intelligible speech at rates as low as 600 bits/s. However,

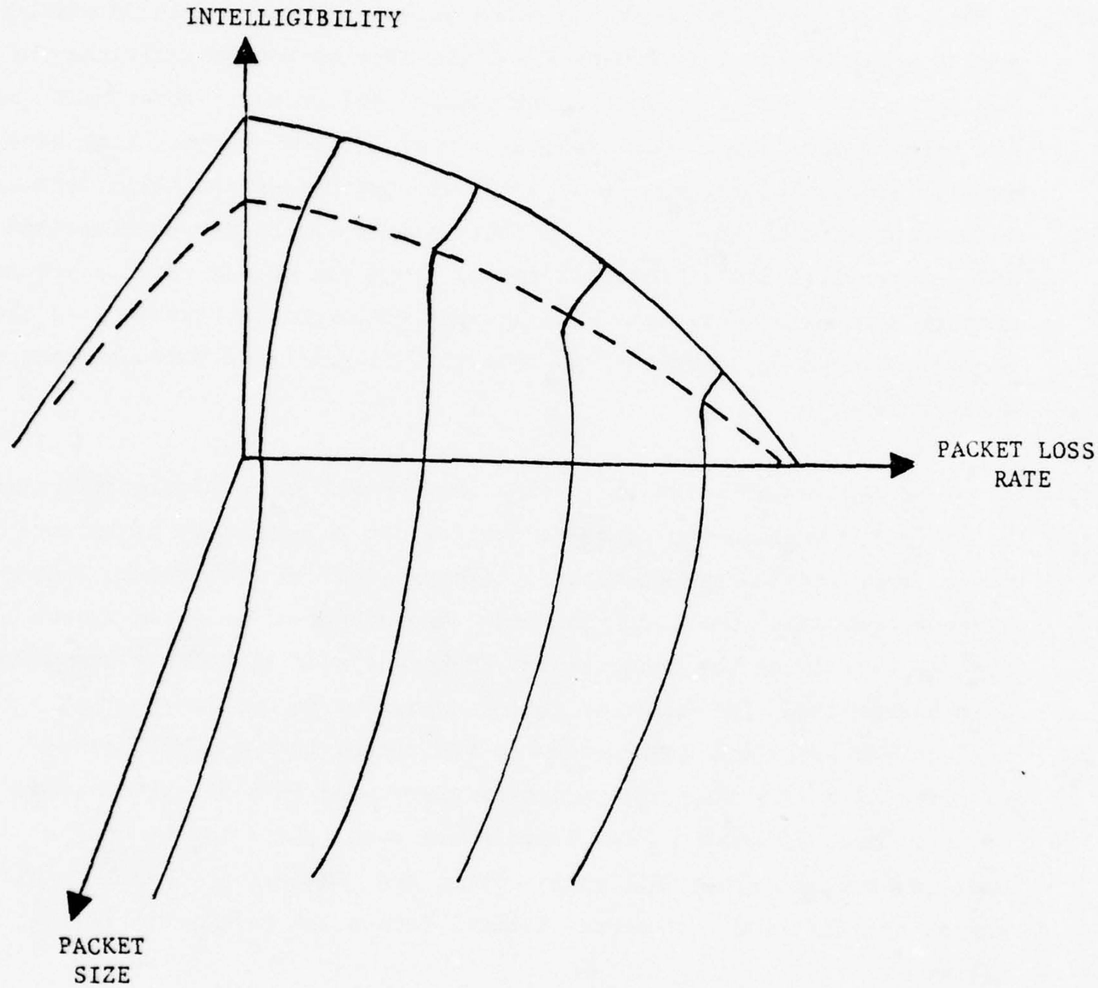


FIGURE 2.2 INTELLIGIBILITY DEGRADATION AS A FUNCTION OF PACKET LOSS RATE AND PACKET SIZE

properties of the speech signal such as the identity and emotional state of the talker are not conveyed well by these systems. The overall naturalness or quality of the speech signal is degraded even in wideband encoding systems such as 16-kbits/s CVSD. Furthermore the efficacy of the systems varies as a function of talker, the acoustic environment of the talker, and so on. For example 3.5-kbits/s LPC encoders that have an MRT intelligibility of 87% at a +26-dB speech-to-noise ratio decrease to 56% intelligibility at a +3-dB SNR, while plain speech decreases only 20% in intelligibility over this range. Thus the choice of vocoders and network strategies must take into account the expected traffic load and the percentages of conversations that will originate in various acoustic environments.

The quality of voice in a noisy environment is particularly crucial to the DCS. Degradation in voice quality can be caused by background noise, poor transducers (handsets), channel-induced bit errors, interference from other users, or jamming. The effect of noise on speech quality depends on the digitization technique used as well as the noise characterisitcs. For example, bursty noise due to overlapping packets will probably be more destructive to PEV speech than to CVSD, since specific bits like the gain parameter correspond to a longer duration of speech. But, if short packet lengths are used, there may be only a small amount of salvageable data. Thus, the characteristics of candidate speech systems in the presence of burst errors and random errors are important.

Finally, it must be noted that no one level of intelligibility or quality can be specified as being proper for all communication. To consider extremes, a one-way message to be played back hours or days later can incur very large temporal distortion, because it will be reconstructed without gaps and can be transmitted at very low bandwidths since there is no real-time requirement. On the other hand, an emergency high priority conversation with one or both particpants highly stressed

and in very noisy environments may require very high throughput rates and very minimal temporal or channel distortion, to be considered adequate. Thus a reasonable analysis of a DAMA satellite system requires either that all voice terminals and network processes be specified at a level sufficiently high to be adequate for the most demanding communication situations or communication situations be categorized and an estimate of the percentage of each kind of category be made.

2.2 VOICE PERFORMANCE EVALUATION

The evaluation of voice performance for various DAMA satellite communication systems depends on many factors. Some of these can be considered independent of the transmission system and others, while dependent can be fixed in order to compare alternative DAMA systems. We first need to separate the DESIGN parameters that can be fixed and the PERFORMANCE parameters that must be varied for each system to give the best performance. This separation is discussed in Section 2.2.1.

One aspect of a DAMA technique trade-off study should be a comparison with existing analog phone systems. Some phone system properties should be retained in future DCS designs. However, digital techniques can allow additional properties advantageous for voice traffic. These are discussed in Section 2.2.2.

The large number of parameters is compounded by the lack of a single voice performance measure. The voice quality measures available and their applicability to the systems of concern in this study are discussed in Section 2.2.3. However, even with sophisticated quantitative quality measures, it is unclear how to make a direct comparison between a transmission system that gives high intelligibility but with a 2-second delay and a system that has low delay but lower intelligibility. An alternative is to define categories of service and compare systems

within the categories. Certainly low-delay and high-delay systems should be in different categories. This is discussed more in Section 2.2.4.

2.2.1 Evaluation of Vocoder Technique Performance

The previous discussion has indicated that there are many parameters related to the overall voice quality and user acceptibility of a voice satellite communication system. For this system study we consider two groups: DESIGN and PERFORMANCE parameters. The design parameters are determined initially and kept fixed for the performance analysis. The performance parameters are varied to meet the performance constraints and to show the parametric effect on the final cost and performance criteria.

These design parameters related to voice communication are considered:

- V - Vocoder type, either PEV (pitch-excited, such as LPC10) or CVSD (continuously variable slope delta modulation).
- b - Bit rate, 4 kbit/s for PEV and 16 kbits/s for CVSD.
- t - Time of speech signal represented by each packet.
- Pe - Channel bit error rate (nominal before coding).
- R - Rate and type of error detection and/or correction.

The performance parameters to be varied so as to give the best performance are:

- Pl - Packet loss rate.
- D - Delay (one-way) for voice transmission.

- Q - The voice quality of the system. The nominal (perfect channel) quality is determined by the vocoder type, bit rate, background noise environment, and audio signal conditioning.

We can give a rough evaluation of the most important parameters for the six access techniques listed below:

1. Preassigned, Not Switched - The only delay for this technique is the fixed propagation delay, which is 0.27 second and generally unnoticeable. Voice quality is a function only of the design parameters; Pe, input SNR, V, and b.

2. Circuit Switched - The same as (1) with some additional delays for call setup. Setup delay does not affect speech quality after the connection has been established.

3. Packet with Infinite Retransmission (This access technique is defined as Category II for one-way voice or data where delay is not a critical parameter) -- In addition to the transmission and queueing delays, there is a variable delay whose distribution is a function of the access parameters, control technique, and system load. This distribution is generally unknown, but as discussed earlier, only the extreme values are significant. Category II service can be described by average delays, since there is no real-time requirement. With infinite retransmission, no channel errors will corrupt the digitized speech, and the nominal quality and intelligibility can be maintained.

4. Packet Without Retransmission - The long two-way delay in a satellite communication system precludes any retransmission if delay is to be unnoticeable. Hence, some packets are received with channel errors and some packets are lost or blocked. The delay with this access technique is like that with the circuit switched, one propagation time, and some buffer queueing. The voice quality is degraded from nominal by the channel bit

errors and the lost packets. For some values of packet time, and hence packet size, this degradation can be reduced. As noted previously, the effect of packet loss rate and packet size (time) on voice quality needs to be parametrically quantified.

5. Packet With Finite Retransmission - At some sacrifice in two-way delay, some of the lost packets can be recovered by finite retransmissions. The large single two-way time of 0.54 second means that the delay distribution will be multimodal with peaks at multiples of 0.27 second. The spread about the peaks will be due to the buffer queueing. The trade-off between increased voice quality and increased delay will be difficult to quantify. Some user effectiveness measure is needed and will be discussed in Section 2.3.

6. Data/Voice Mix [of (3) and (4)] - Adding data to the voice traffic increases the probability of packet collisions. The lost-packet rate increases and voice quality decreases. The effect of increasing the mix will thus be reflected in lower voice quality.

2.2.2 Comparison of Analog and Digital Voice Communication Systems

The evaluation of a voice transmission system must include user acceptability and speech quality measures. Acceptance cirtieria for data transmission are usually bounds on error rates and average delays. In a voice system, these criteria do not sufficiently describe all potential degradations from standard voice communication systems such as the phone system. In designing analog voice networks, it has been common to assume that ordinary human telephonic conversation can be classified as:

1. Full Duplex - Two people should be able to talk at the same time and each should be able to hear both his own and the other person's speech.

2. Fixed Fidelity - The quality of a given conversation is a function of the voice transducers (e.g., the Western Electric carbon button handset) and of the transmission bandwidth and SNR. The quality of voice may vary during a conversation, but this is considered to be a fault of the transmission network, and generally correctable by establishing a new connection. In general, variation in quality is not expected during a local connection in the United States. Changes in quality may also occur due to dirt and compaction in a handset, but these are very long term and again are considered aberrations in the system to be fixed by repair or replacement of a handset.

3. Minimal Delay - When a connection has been established, it has been expected that the delay between talking and reception would be very short, compared to human response times, i.e., less than 20 ms. Even with satellite links that introduce around 300 ms of delay, two-way conversation has not been impaired.

In early digital communication systems, these same assumptions have been made. That is, both people should be able to talk and listen simultaneously; the quality is fixed by the kind of voice digitization and the bit rate of transmission, a given bit error rate for the channel, and so forth. It is not surprising that these assumptions, which are in accord with early technological capabilities, make performance analyses of circuit-switched networks look much better for speech communication than do those of packet switched networks.

However, the above characterization of conversations and of speech ignores several advantages of digital communication systems over analog

communication systems and also ignores potential advantages of packet-switched systems over circuit-switched systems such as the following:

1. All segments of speech do not require the same amount of coding; silence requires many fewer bits than a voiced-consonant/vowel transition.
2. The most recent segments of speech are often highly predictive of the current segment to be coded, from the very micro level up to very macro levels.
3. For a given set of channel resources, there are trade-offs between different kinds of degradation; the optimal trade-off will be a function of the user-pair; their task, their environment, and so forth.
4. It may be desirable to allocate different channel resources to different users, according to priority, acoustical environment, and the like.
5. It is possible to build voice digitization equipment (VDE) that degrades gracefully as resources are lowered, and thus the possibility arises of intelligent networks that can always keep traffic flow optimal and voice quality maximal.

2.2.3 Quantitative Measures of User Acceptibility and Speech Quality

Many voice digitization techniques when tested, give intelligibility scores equivalent to those of the phone system but still differ from that reference. The discrepancy is often referred to as a quality difference. Currently, researchers are attempting to define a quantitative measure of quality that are consistant and repeatible but do not correlate strongly with intelligibility scores. Recently, it has been found that talker recognizability does not correlate with intelligibility, but does correspond to subjective rankings of voice digitization techniques with respect to preservation of talker identification clues. Such tests could be useful in comparing quality and user acceptibility for the candidate systems.

Another quality measure that gives results not predictable from intelligibility scores is a preference measure called paired acceptibility ranking method (PARM) [72]. This measure, along with DRT intelligibility scores, was used in an extensive series of voice digitization tests monitored by the DOD narrowband consortium. When these data become available, they will be very useful for comparison of voice digitization equipment.

In order to investigate the effects of temporal degradation on user acceptance and user performance efficiency in two-way communication, a test facility for conducting two-way communication tasks over a simulated network is used in this study. Specified degrees of temporal and digital encoding degradation can be introduced under program control. By measuring task performance efficiency and changes in verbal behavior, the effects of various system configurations on the ability of people to communicate effectively can be determined.

The literature uses speech quality rather loosely to mean many things. In the remainder of Section 2, speech quality is used as a term denoting the results of the specific tests cited above.

2.2.4 Categorization of Voice Service

The effectiveness of a user of a voice communication channel is maximized when the voice quality is high and the delay is low. User effectiveness for the total system is also dependent on the blockage probability. Thus, user effectiveness is a function of several parameters that cannot be compared directly. For instance, RMA access techniques have no blocking but do cause large delays if a retransmission strategy is used. The two parameters of delay and packet time give four types of voice service (see Figure 2.3):

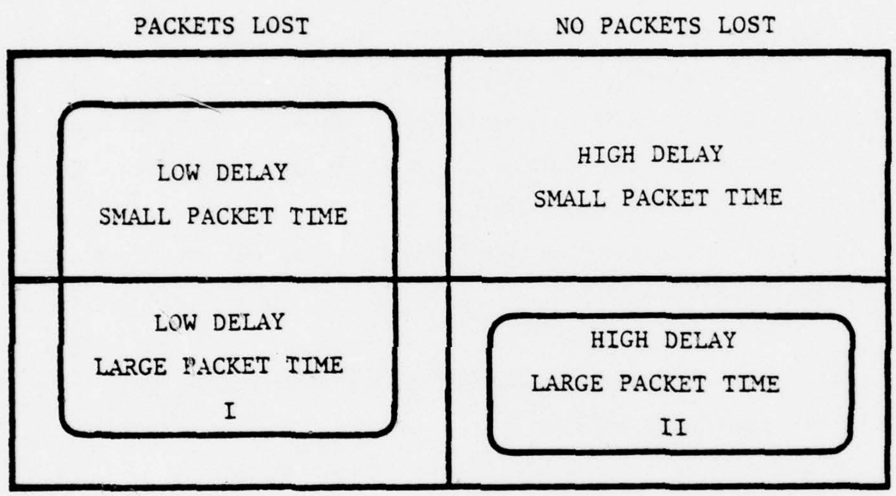


FIGURE 2.3 CATEGORIES OF VOICE SERVICE

1. Low delay, Small packet time - The small packet time allows the receiver to smooth over lost packets (Pl = 5 -15%).

2. High delay, Large packet time - This type is best used for one-way messages or other service where speech fidelity is more important than delay. Pl can be essentially 0.

3. Low delay, Large packet time - Some users may prefer increased packet time, so that they know when the channel deletes a packet. Large packet times increase bandwidth utilization efficiency but cause very apparent gaps. Voice quality is very difficult to quantify.

4. High delay, Small packet time - This is a logical but not useful type. The high delay would allow low Pl and thus there is no need for small packet times.

We can group these types into two categories for the cost analysis (Figure 2.3):

1. Low delay, Small packet time - This includes types (1) and (3) since (1) is an upper bound for (3).

2. High delay, Large packet time - This is the service used for data, one-way voice messages, and voice communication with high priority on fidelity.

2.3 BASELINE SYSTEM PARAMETER ANALYSIS AND COST FACTORS

For this analysis to be tractable, it is necessary to limit the voice digitization study areas. The intention of this study is to show the feasibility of DAMA techniques for a common user network with satellite transmission. It is beyond the scope of this work to study exhaustively all possible vocoder variations and other parameter variations. We recognize that these matters will have an impact on future DCS design, but cannot possibly do justice to them here.

The cost factors associated with voice digitization equipment (VDE) are discussed in Section 2.3.1. Analysis of the current trends in LSI technology indicate that this cost is dominated by the other earth terminal costs, especially for wideband VDE. The relationship between delay and packet loss rate is discussed in Section 2.3.2. More retransmissions decrease Pl but increase delay. Thus user effectiveness in two-way communication tasks is affected. Data-bit-error rate requirements will dominate voice requirements, and so no trade-off of Pe and cost is needed. Pe will affect the packet loss rate, however, in that packet headers will be in error. Section 2.3.3 discusses the use of error detecting and correcting codes to protect packet headers.

2.3.1 Cost Factors Associated with Voice Digitization Equipment

The assumptions of this study relevant to costing of VDE are that:

1. VDEs will be located at nodes or earth terminals (ET).
2. Input to the VDEs will be passed through local PBXs with standard (carbon button) handsets.
3. The same VDE will be used at all nodes.
4. No tandeming of different VDE will be considered.
5. Two baseline types of VDE will be used in the performance analysis, 16-kbits/s CVSD, and 4-kbits/s PEV.

Wideband CVSD VDE is very inexpensive and clearly will not affect the earth terminal costs. Low-rate (PEV) techniques can be implemented on specialized programmable CPU-based hardware costing about $2000. Since these costs are mainly for the memory, some cost reduction can be achieved by pooling memory at nodes where several voice channels exist. If the DARPA vocoder-terminal program achieves its goals, the cost cost of PEVs will be small enough so as not to affect the earth terminal costs. Thus we will not consider voice digitization cost as part of the earth terminal costs.

2.3.2 Delay and Reliability Trade-Offs

For the satellite wide-bandwidth systems under study, the delay can be fixed and is a characteristic like blocking probability. For zero retransmissions, a one-way delay of 0.3 second is virtually unnoticeable, one retransmission gives 0.6 second delay and may be noticeable, two retransmissions gives a delay of 0.9 second which is surely noticeable and will affect the communication. The effects of delays in the range of 0.3 to 1.08 second can be studied by using a two-way communication task experiment. The experiment is designed to quantify the amount of information transfer over a communication channel with fixed delays and packet losses. Section 2.3.4 describes the methodology of such an experiment and how the results can be used in evaluating system design trade-offs.

Another characteristic of DAMA systems that is important for voice is whether the assigned channels can be reassigned during nonspeech periods. Packet systems have this advantage and thus require about half the nominal bandwdith of fixed assignment systems. Also, the non= stationary nature of voice signals enables another reduction if the DAMA system allows asynchronous rates (variable length packets). The variable-rate strategy referred to before as DELCO gives an average reduction in transmitted bit rate of 60% for a 4-kbits/s vocoder. This is in addition to the 50% speech activity reduction. The quality is comparable to a 4-kbits/s systems with an average rate of 2.4 kbits/s.

TABLE 2.2 VOICE DESIGN PARAMETERS

Type	Cost	Max. No. of Retrans.	Nominal Bit Rate
CVSD	$5-100	2	16 kbits/s
PEV	$2000-$8000	2	4 kbits/s

2.3.3 Voice Quality Degradation Due to Packet Loss and Bit Error Rate

The channel bit error rate (Pe) that can be tolerated without additional quality degradation is 1% for CVSD, and 0.1% for PEV in systems without packet loss. For packet systems, the effect of Pe on packet loss rates (Pl) due to corrupted headers must be considered. The somewhat conservative goal for total Pl is 5%. Assuming that the average number of channels active at any one time is 8000, a 13-bit header index is necessary to reference all users. Two types of error coding are considered, 4-bit parity-error detection and 13-bit forward-error correction. The efficiency of the bandwidth utilization is a function of the packet time (set to 20 ms) in order to maintain voice quality for 5% Pl. (Packet time is the amount of speech in milliseconds that is represented in each packet.) Table 2.3 summarizes the interaction between Pe and Pl for three systems.

The cPl (channel packet loss rate) is due to detected errors in the headers. The aPl (access Pl) can then be used to design the packet access system and associated bandwidth.

The Pl of 5% is approximately the point at which speech quality begins to degrade. It is believed that intelligibility remains fairly constant to Pl of 15% but that the choppiness introduced becomes noticeable at 5% and objectionable at 15%.

2.3.4 Two-Way Communication Experiment Results

In the preceding discussion it has been noted that it is possible to make highly reliable, repeatable measurements of the intelligibility of a particular communication system, provided that proper precautions are taken in the laboratory administration of such tests. It has also been indicated that reliable measurements can be made of transmission of

TABLE 2.3 PE AND P1 DESIGN PARAMETERS

Vocoder Type	Data Bit rate (bits/sec)	Header Length (bits)	Packet Length (bits)	Total Bit rate (bits/sec)	Pe (%)	cPl (%)	aPl (%)
1. Non Packet Loss System							
CVSD	16K	-	-	16K	1	--	--
PEV	4K	-	-	4K	0.1	--	--
2. Packet Loss System with 4-Bit Error Detection							
CVSD	16K	13 + 4	337	16,850	0.01	0.15	4.85
PEV	4K	13 + 4	97	4,850	0.01	0.15	4.85
3. Packet Loss System with 13-Bit Error Correction							
CVSD	16K	13 + 13	346	17,300	1	0.01	5.0
PEV	4K	13 + 13	106	5,300	0.1	0.00	5.0

other extra-intelligibility voice attributes, such as talker identification. It is also possible to determine reliably whether some particular user population will significantly prefer to use one of the candidate systems. That is, a particular communication method can be described by several different test results, and the importance of each result will depend on the way in which the system is going to be used, the kinds of users, the tasks to be performed, the acoustic environments in which the system will be used, and so on.

However, the fact that reliable measurements can be made does not indicate how to use them to make a cost effective decision in designing an overall system. There is no known wasy to combine the information that System A is 3% more intelligible than System B, or that System A is preferred by 80% of users, with the fact that System A is 30% more expensive or uses 20% more bandwidth.

To help make such design decisions, a two-way communication task, described in Appendix A and shown in Figure 2.4, is used to determine how well two people can use a candidate system to convey information to one another. Such results are not easily derivable from standard intelligibility and quality ratings. The concept of conversational effectiveness has been considered in intelligibility testing for a long time, beginning with Fletcher [210]. Another example of conversational measurement is the free conversation test [Richards, 211]. In particular, we are attempting to measure information exchange in bits per second for various tasks as a function of system parameters.

The value of this approach is that while it is implicitly dependent on such factors as intelligibility, it also takes into account the effects of such distortion factors as temporal delay. Furthermore, it yields numeric results that are more easily combined with cost and bandwidth requirements in making design decisions. This method has been developed for assessment of various packet speech transmission methods

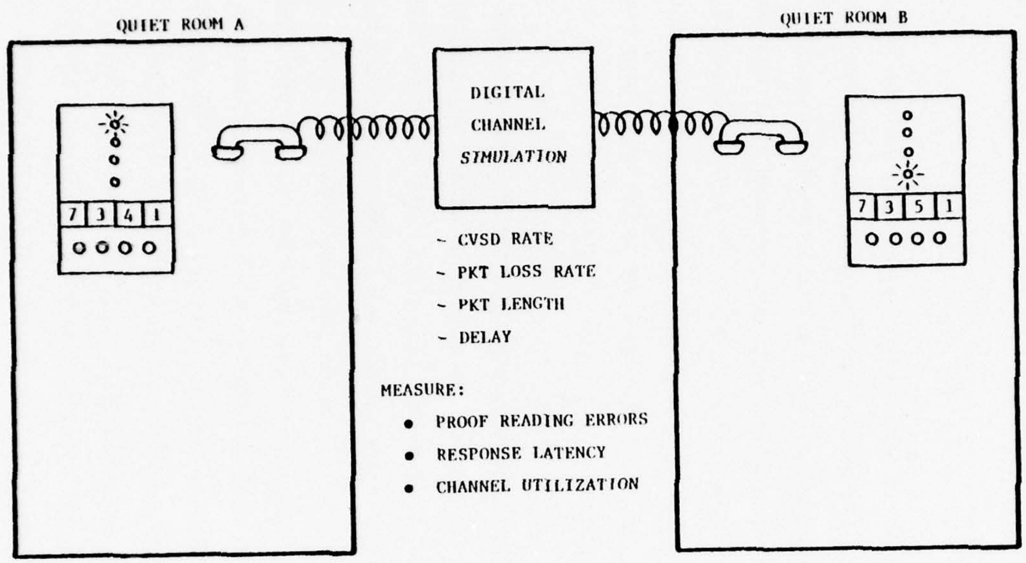


FIGURE 2.4 TWO-WAY INTERACTIVE TASK COMMUNICATION EFFECTIVENESS EXPERIMENT

under the sponsorship of the Defense Advanced Research Projects Agency (DARPA). A pilot study was initiated under this contract, using these methods in order to develop a methodology for evaluating the communication effectiveness of the candidate demand access techniques. The results discussed here show the effects of varying specific parameters related to satellite packet systems: fixed delay, packet loss rate, packet length, and CVSD rate.

The results of this experiment are shown in the next four figures. In each figure, the effective information transfer rate is shown in bits per second as a function of one of the system parameters. For reference, the performance is also shown for two systems that are expected to be quite good and very bad respectively. In each figure the results are shown for the task where both people must have the correct answer before they can continue, and for the case in which only one person must have the correct information.

Figure 2.5 shows the effects of introducing various degrees of fixed delay into a channel. As would be expected, increasing delay decreases information transfer for this kind of task. In particular, for the case in which both people must have the correct information, an increase of delay from 270 ms to 540 ms decreases information flow from 3.2 bits/s to 2.78 bits/s. Thus, one can expect the transfer of N bits of information over such a channel to take 1.15 times as long for this kind of task in a 540 ms delay channel as in a 270 ms channel. Similarly, increasing the delay to 1080 ms will require 1.4 times as long to transfer the same amount of information. This illustrates an obvious case in which the parameter of interest does not show any effect in intelligibility scores, talker recognition scores or preference scores in one-way communication tests and yet suggests that the 1080 ms delay may increase total load on a network by 40%.

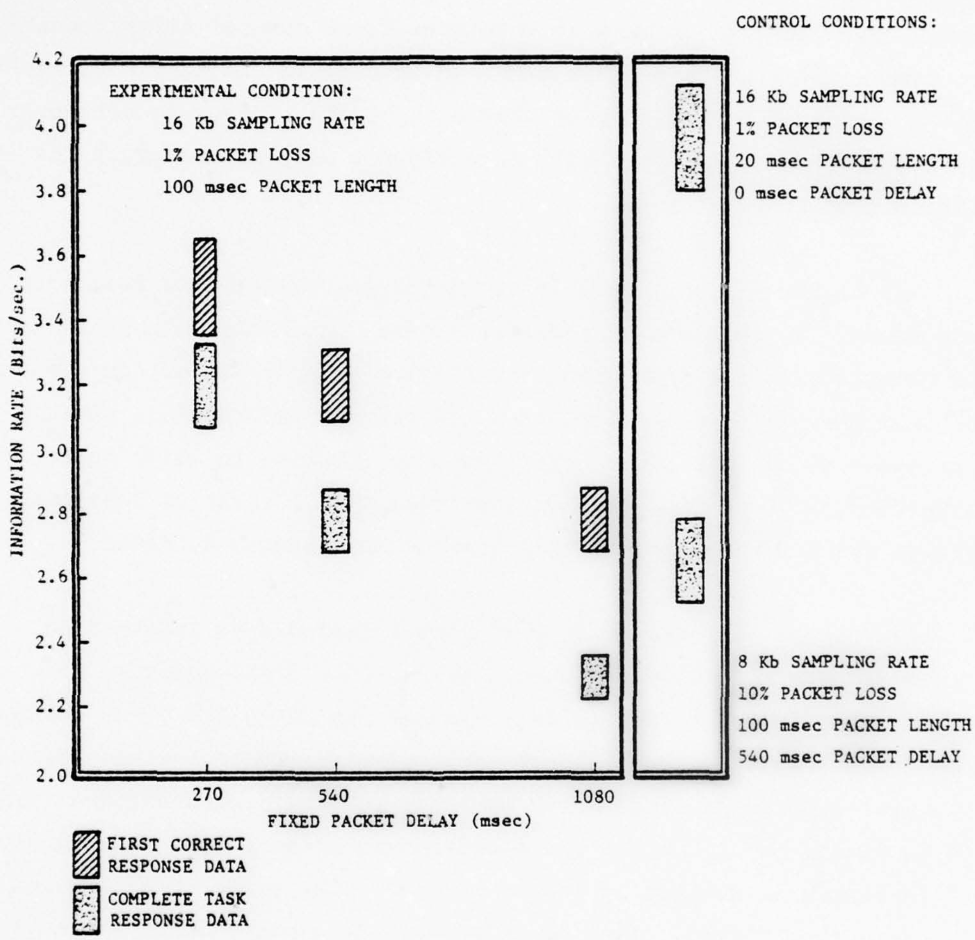


FIGURE 2.5 INFORMATION TRANSFER AS A FUNCTION OF FIXED PACKET DELAY

Figure 2.6 presents the same information in a different way to illustrate how the information transfer rate was derived. The ordinate in this case is the latency time for both subjects to make a correct response. Thus, the effect of increased fixed channel delay (caused by retransmissions, say) on task response latency is directly observable. The number of information bits that must be transferred to correctly complete the task are used with this latency measure to derive the information transfer rate.

In Figure 2.7, the results of increasing packet loss from 1% to 10% are shown. In the case of complete two-way interaction, this increase in packet loss rate increases conversation time by 6%, or, in the case of only one person needing to have the correct information, time is increased by 3%. This case represents an instance in which quality is significantly worse for the 10% loss case than for the 1% loss case, and yet in which performance in this task is not greatly hindered.

In Figure 2.8, the effects of packet length upon information transfer rate are shown. The results for this task show that the information transfer rate is very similar for 20-ms and 50-ms duration packets and thus suggest the use of longer packets to increase efficiency. However, packets as long as 100 ms increase total conversational time by 6% to 8% and may result in increased conversational load that negates the overhead advantages of longer packets. The shape of the curve also suggests that further experiments should be conducted with packet lengths up to 500 ms.

In Figure 2.9, the effect of different bit rates on information transfer rate are shown. Here it can be seen that decreasing the bit rate of the CVSD speech from 16 kbits/s to 8 kbits/s decreases conversational throughput by only 4%. This case is particularly interesting because previous research has indicated that equally intelligible systems

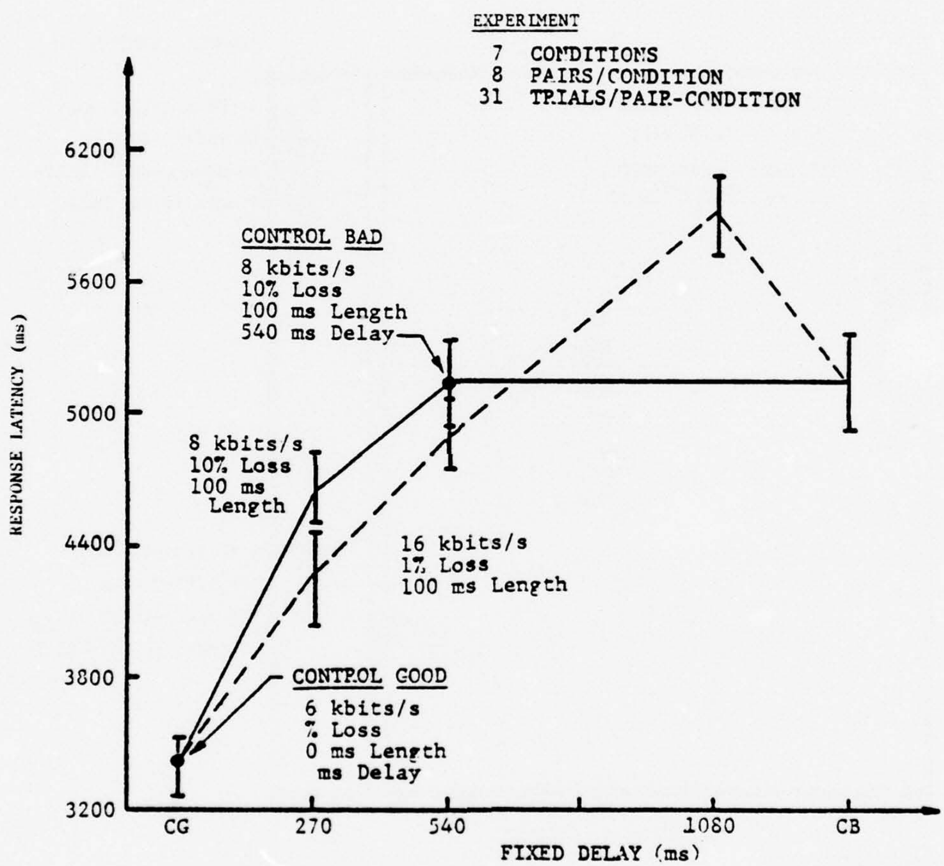


FIGURE 2.6 CORRECT RESPONSE LATENCY AS A FUNCTION OF FIXED PACEKT DELAY

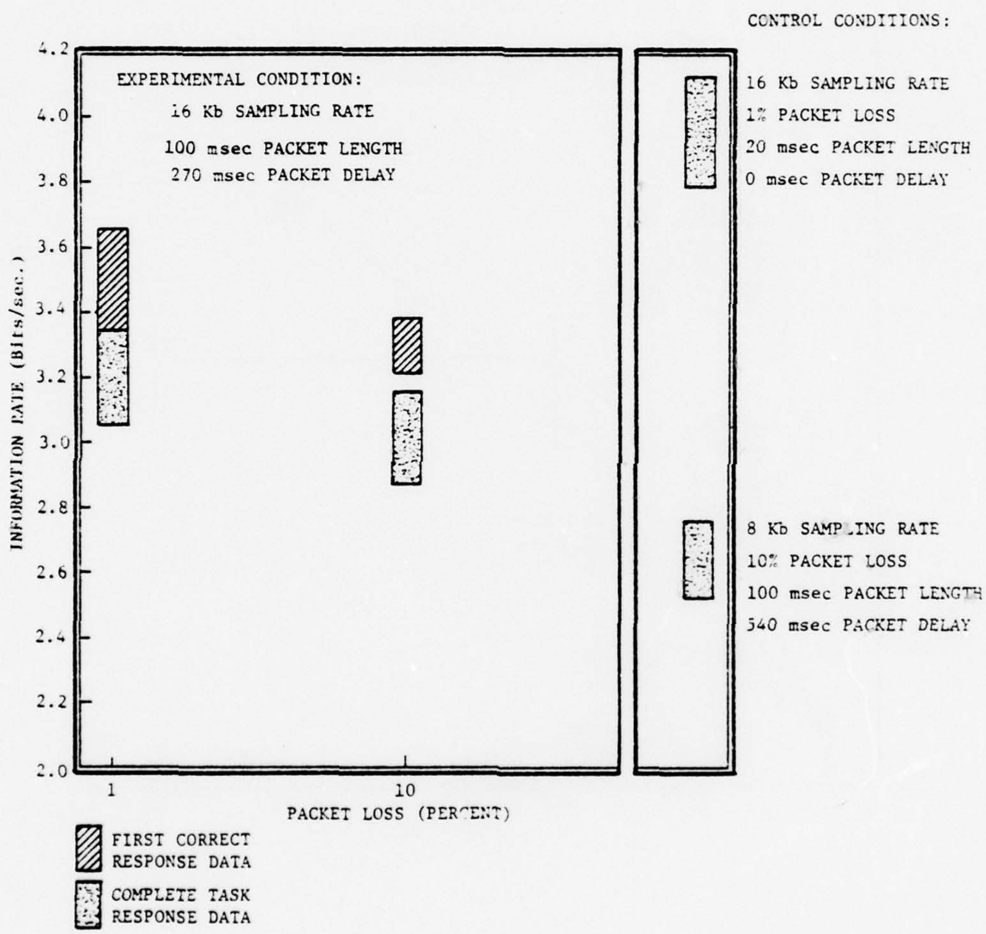


FIGURE 2.7 INFORMATION TRANSFER AS A FUNCTION OF PACKET LOSS

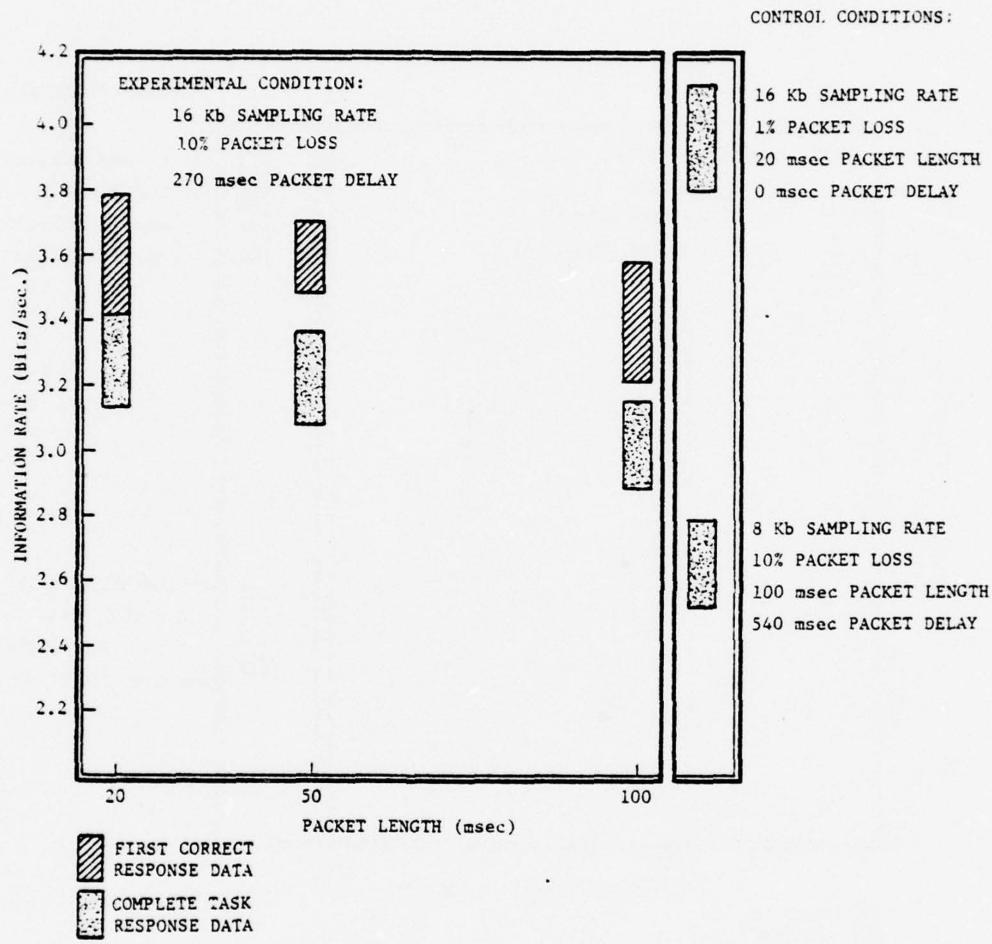


FIGURE 2.8 INFORMATION TRANSFER AS A FUNCTION OF PACKET LENGTH

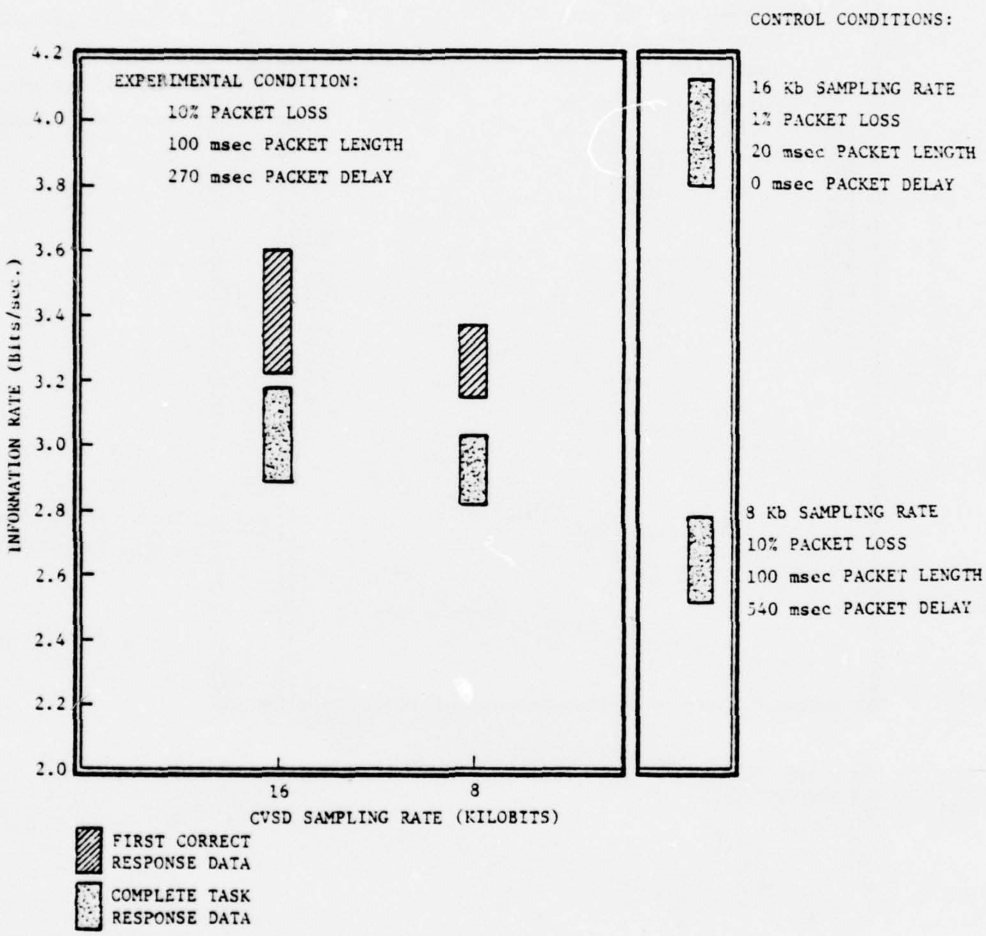


FIGURE 2.9 INFORMATION TRANSFER AS A FUNCTION OF CVSD RATE

may vary significantly in other dimensions of speech quality. This experiment, however, indicates that a system that is significantly better in intelligibility and preference may be only marginally more effective in actual transfer of information in this kind of task.

These results can be incorporated with the channel utilization analysis for RMA packet techniques to study some design trade-offs. Table 2.4 shows a summary of the channel utilization percentages for pure, slotted, and capture ALOHA techniques. Various ratios are computed in order to compare utilization imrpovements and communication improvements and communication effectiveness reductions for the various values of system design parameters.

TABLE 2.4 CHANNEL UTILIZATION AND COMMUNICATION EFFECTIVENESS FOR VARIOUS RMA SYSTEM VALUES

	Information Transfer Rate		ALOHA Method								
			Pure			Slotted			Capture		
Loss rate -	1%	10%	1%	5%	10%	1%	5%	10%	1%	5%	10%
Number of Retransmissions											
0 (270 ms)	3.2	3.0	0.005	0.026	0.053	0.010	0.051	0.110	0.030	0.100	0.210
1(540 ms)	2.78	-	0.048	0.105	0.145	0.096	0.210	0.290	0.177	0.450	0.616
2 (810 ms)	2.5	-	0.075	0.150	0.85	0.150	0.300	0.370	0.283	0.630	0.844
Ratios of Above ------											
1% to 10%	1.07		10.0			11.0			7.0		
(270 ms)											
0 to 1	1.15	-	9.60	4.04	2.73	9.60	4.04	2.73	5.90	4.50	2.93
1 to 2	1.11	-	1.57	1.43	1.27	1.57	1.43	1.27	1.63	1.40	1.37

This table indicates a few things about the two-way communiation task. First, the simple proof-reading task does not sufficiently expose the known differences in degraded channels, such as going from 1% packet loss to 10% loss. A more complicated task with higher-information-content words is needed. However, the ratio of channel utilization for one to two retransmissions is not significantly higher than the information rate loss, indicating that, even for this simple task, more one retransmission should not be used for voice communication systems.

Another measure that should be used is the channel activity ratio. During the experiments described, measures of speech activity were made, and data were collected. Sufficient controls were not provided to properly correct for false starts and equipment failures. Thus, the data cannot be used to determine whether additional channel usage occurred to conpensate for degradation such as higher packet loss rates. A course review of the data indicated that this was not the case, however, which says that the analysis from Table 2.4 is legitimate.

In summary, any voice digitization method can be described by a variety of measures; these measures will sometimes (but not always) be relevant to the decision of the communications engineer. It does appear from this pilot study that communication system design must take into account the ability of two people to transmit information between each other, both for problem solving (as in this experiment) and for other types of intercourse, such as political debates. Future experiments should include both types of tasks, in addition to being more taxing so as to clearly show the effects of channel degradation. One other observation is that possibly, quality should be emphasized over intelligibility in designing voice communication systems. That is, given a certain minimal level of intelligibility, user acceptance will be most influenced by the quality (how well it sounds) and the task performance capability (how well it works).

3 COMMON USER NETWORK/TRAFFIC MODEL

In this study, a baseline network traffic model is necessary to provide a cost comparison basis for the tradeoff analyses between an all-terrestrial system and various satellite DAMA/terrestrial system configurations. This chapter describes the common user traffic model, its baseline, all terrestrial network models, and baseline terrestrial costs used for this study. These models were derived based on the projected DCA traffic and network design considerations data for the late 1980's time frame in the contract SOW and other related reference models [59].

3.1 USER TRAFFIC MODEL

The user community is described in terms of DoD or military installations, categorized in terms of their gross size. An individual location would not normally be an *actual DoD location*, but some location where a number of DoD personnel may be situated. The number of each location type, together with its characteristics in terms of staffing and communications traffic is shown in Table 3.1.1. A more detailed classification of the data communications sources is shown in Table 3.1.2. Further refinement of the data traffic generated per terminal during the busy hour is shown in Table 3.1.3 for high and low speed terminals and computers.

The voice traffic shown is specified in terms of total originating traffic. It is assumed that on the average the users of a voice terminal originate 3.6 CCS of traffic during the busy hour. However, approximately 80% of this traffic is destined to users within the same location as the originator and is thus not seen by the DCS as it is "turned around" at the local PBX level. It should be assumed that by the late 1980's between 10% and 20% of the voice terminals will be secure voice terminals, and thus inherently digital with ciphered bit streams.

TABLE 3.1.1 ESTIMATED WORLDWIDE VOICE AND DATA TRAFFIC - LATE 1980'S

	Large	Medium	Small	Very Small	Individual	Totals
Number of Locations of each Type, Worldwide	140	350	490	770	770	2520
Average Staffing	11300	2116	211	34	3	
Average Number of Voice Terminals per Location (1)	1835	346	34	6	1	
Aggregate Number of Terminals	256900	121100	16600	4620	770	400050
Average Originating Busy Hour Voice Traffic (Erlangs) (2)	37	6.92	0.68	0.12	0.1	
Aggregate Traffic - Originating Erlangs	5180	2422	333.2	92.4	77.0	8104.6
Originating Busy Hour Data Traffic - all sources in bits/hour per location	103.5×10^6	33.7×10^6	8.83×10^6	1.52×10^6	0.418×10^6	

(1) For purposes of this study assume that between 10% to 20% of the voice terminals are secure voice terminals.

(2) Based on 3.6 CCS per terminal in busy hour and 80% of the traffic turned around at the base level PBX. Thus only 20% of the traffic actually reaches the DCS.

TABLE 3.1.2 ESTIMATED DATA TRAFFIC PER LOCATION

	Large	Medium	Small	Verv Small	Individual
On-Line Data Users Per Location					
Computers	10	2.5	0.5	--	--
Low Speed Terminals	34	19	7	2	1*
High Speed Terminals	16	9	3	1	--
LDMX/Narrative	1	1	1	1	--
On-Line Data Traffic - Per User-KB/Busy Hour					
Computers	7870 KB	7870 KB	7870 KB	--	--
Low Speed Terminals	396 KB	396 KB	396 KB	396 KB	396 KB
High Speed Terminals	702 KB	702 KB	702 KB	702 KB	--
LDMX/Narrative	129.6 KB	129.6 KB	21.6 KB	21.6 KB	21.6 KB
Aggregate Data Traffic - Per Location KB/Busy Hour					
Computer	78700 KB	19700 KB	3935 KB	--	--
Low Speed Terminals	13480 KB	7520 KB	2770 KB	792 KB	396 KB
High Speed Terminals	11220 KB	6320 KB	2106 KB	702 KB	--
LDMX/Narrative	130 KB	130 KB	22 KB	22 KB	22 KB
Total - per location	103,530 KB	33,670 KB	8,830 KB	1,516 KB	418 KB

*Single terminal used for both data and narrative.

TABLE 3.1.3 DATA TRAFFIC GENERATED PER TERMINAL DURING BUSY HOUR

User Mode	Data Rates	One-Way Transaction Time	No. of One-Way Transaction Per Busy Hour	Average Information Bits Aggregated
Data Terminals	Low Speed 150 to 600 b/s 450 b/s nominal	Query 4 sec	20	36 KB
		Response 40 sec	20	360 KB
	High Speed 2.4 kb/s to 4.8 kb/s 3.6 kb/s nominal	15 sec	13	702 KB
Computer-to-Computer				
Query	Each computer uses a nominal 4.8 kb/s network access rate	3 sec	130	187.2 KB
Response		4 sec	175	3.36 MB
Bulk (1)*		60 sec	15	4.32 MB
Bulk (2)**		1 hour	2 (per day)	34.56 MB

* Bulk (1) - short file transfer or data base revisions.

** Bulk (2) - long file transfer, major data base transfers, etc. Normally handled outside of busy hour.

The amount of data traffic shown in Tables 3.1.1 and 3.1.2 makes up only approximately 5% of the total DCS user traffic (assuming 16 kbps voice encoding rate). Thus, for capacity and system cost purposes, the data portion of the traffic can effectively be ignored. It is important to point out, however, that all candidate DAMA systems should be designed so that data traffic can be processed efficiently and economically to meet the user requirement.

The traffic model can be summarized by the following parameters:

Totals: Total Terminations $\cong 4 \times 10^5$
Busy Hour Traffic = 0.1 Erl. (3.6 ccs)/term
Inter-PBX Factor = 0.2 (3.1)

Based on the above, the total busy hour originating voice traffic which is switched and potentially carriable by the satellite system is:

$$E = (0.2)\ (0.1)\ (4 \times 10^5) = 8000 \text{ Erl.}$$

For purposes of this study we shall make some idealized assumptions* on the geographic distribution of origins and destinations of the terminal areas (since each is to be associated with an earth terminal). By definition of terminal areas, the traffic from each terminal area E_i, $i = 1, 2, \ldots, N$, sums to the total originating traffic E,

$$E = \sum_{i=1}^{N} E_i \qquad (3.2)$$

---

* For purposes of overall comparison for sizing costs and different access access techniques this idealization is not, in our judgment, significant. For specific design purposes, such traffic distribution characteristics vary in importance with DAMA techniques. Such sensitivities are described in the DAMA concept chapter of this report.

The fundamental idealized traffic flow assumption to be made is that the termination of traffic originating in one terminal area is divided among all terminal areas according to their relative originating traffic. Expressed as an equation:

$$E_{ij} = E_i E_j / E \tag{3.3}$$

This model will be called the balanced-flow assumption. Let T_j represent traffic terminating in area j, i.e.,

$$T_j \triangleq \sum_{i=1}^{N} E_{ij} \tag{3.4}$$

Then the balanced flow assumption gives:

$$T_j = E_j \tag{3.5}$$

Hence, a first consequence of the balanced flow assumption is that every terminal area terminates exactly as much traffic as it originates.

From (3.5) we also deduce that the traffic originating in a terminal area but not leaving is just

$$E_{ii} = E_i^2 / E \tag{3.6}$$

and the traffic originating in area i and outgoing, defined as:

$$E_{i\text{-out}} = \sum_{\substack{j=1 \\ j \neq i}}^{N} E_{ij} \tag{3.7}$$

is given by

$$E_{i\text{-out}} = E_i \left(\frac{E-E_i}{E}\right) \tag{3.8}$$

Similarly, it follows that traffic incoming to area i and terminating there is given by:

$$E_{i\text{-in}} = E_i \left(\frac{E-E_i}{E}\right) \tag{3.9}$$

Not mentioned thus far is the tandem traffic for a terminal area which, for the terrestrial switched network, originates outside an area and terminates outside the area but uses tandem switching to cross it. It is clear that the tandem traffic issue depends highly on the topology and routing plan of the network. Hence, we shall introduce the terrestrial network model before the tandem traffic model.

3.2 BASELINE TERRESTRIAL NETWORK MODEL

This model follows the basic DAMA traffic model as described previously, and provides a simple and realistic estimate of the terrestrial trunking cost savings for different satellite terminal configurations. This model is derived by engineering terrestrial switched networks for three different satellite terminal configurations and curve fitting the three systems with a simple and general expression (total terrestrial trunk-miles in terms of number of earth terminals) that can be readily computed for further cost saving analysis.

The baseline system is derived from the network design considerations discussed in [59] and the traffic model from the previous section. The baseline system is assumed to be an all-terrestrial switched network in the CONUS area with 70 switches and 149 trunk groups. The topology of the network is of the modified hierarchical network concept with a hierarchical structure (seven regional switches) overlaid on a

distributed polygrid network. Each regional switch provides most of the inter-regional communications for a community of ten switches and direct trunk groups are provided for all local switch to regional switch connections. The seven regional switches are fully connected by 21 trunk groups; hence, all inter-regional calls can generally be served with three or fewer trunks (except alternate routing or switch outages). The average length of an inter-regional trunk as measured in [59] is 1,250 miles. The distributed portion of the network is assumed to be a polygrid network with an average of four trunk group connections per switch (follows from the 149 trunk groups). Hence, for each switch, this gives direct connections to four switches, 2-trunk connections to eight switches (from the 4-polygrid assumption), and 3-trunk connections to the remaining 57 switches (from the hierarchical network assumption). The average trunk length in the distributed network in [59] is 260 miles. All trunk groups in the network are assumed to have a 92% busy hour occupancy.

From (3.2) the total originating traffic in the network is:

$$E = \sum_{ij} E_{ij} \text{ Erlangs,} \tag{3.10}$$

where E_{ij} is the originating traffic from switch i to switch j.

a. Fully Connected Network

The total traffic carried in the fully connected regional network (assuming equal traffic case so that $E_{ij} = \frac{E}{70^2}$ for all i, j) is:

$$\sum_{\text{all switches}} \left\{ \sum_{\substack{\text{all switches}\\ \text{that cannot be}\\ \text{reached by two}\\ \text{trunks or less}}} \left[E_{ij} = 70 \times 57 \times \frac{E}{70^2} \right] \right\} = \frac{57E}{70} \text{ Erlangs} \tag{3.11}$$

With one trunk per call and an occupancy factor of 0.92, the total number of trunks in the fully connected network is:

$$\frac{57E}{70 \times 0.92} = \underline{0.88 \text{ E trunks}}, \text{ or } \underline{1100 \text{ E trunk-miles}} \qquad (3.12)$$

b. Distributed Network

The total traffic carried in the distributed network in Trunk-Erlangs is non-inter-regional traffic + inter-regional traffic

= ((total traffic using 1 trunk) x 1 trunk + (total traffic using 2 trunks) x 2 trunks)

+ (inter-regional traffic) x 2 trunks

$$= 70 \times 4 \times \frac{E}{70^2} \times 1 + 70 \times 8 \times \frac{E}{70^2} \times 2 + \frac{57}{70} E \times 2$$

$$= 1.914 \text{ Trunk-Erlangs} \qquad (3.13)$$

With an occupancy factor of 0.92 and assuming evenly distributed load, the total number of trunks in the distributed network is:

$$\frac{1.914E}{0.92} = \underline{2.08 \text{ E trunks}}, \text{ or } \underline{540 \text{ E trunk-miles}} \qquad (3.14)$$

Therefore, in all terrestrial systems (N=1) the total terrestrial network is 540 E + 1100 E = 1640 E trunk-miles. In the case where all inter-regional networks are replaced by satellite (N=7) the fully connected regional network is eliminated and the total terrestrial network is 540 E trunk-miles.

The third case considered is a system of 35 satellite terminals (N=35). Only 35 trunk groups are required, each connecting a pair of local switches and the satellite terminal is collocated with one of the two switches. The average trunk length here is 170 miles.

The total traffic in the terrestrial network is:

35 x (traffic between the 2 switches + remaining inter-switch traffic for 1 of the switches)

$$= 35 \times (2 \times \frac{E}{70^2} + 2 \times \frac{68}{70^2} \times E) = 0.985E \text{ trunk-erlangs} \tag{3.15}$$

Assuming 0.92 occupancy and 170 miles/trunk, the total terrestrial network is:

$$\frac{0.985E}{0.92} \times 170 = 182\ E\ \underline{\text{trunk-miles}} \tag{3.16}$$

Finally, at N=70 (one terminal/switch) all interswitch trunking is eliminated. By assuming linear interpolation between the four canonical cases (N = 1, 7, 35, 70) we have derived a functional relationship between terrestrial trunk-miles and N, the number of earth terminals. This function is plotted in Figure 3.2.1 as a percentage saved of the all-terrestrial base trunk-milage. This function is converted into annual cost savings in Section 3.3.

c. Tandem-Switched Traffic

Let k be the average number of terrestrial tandem switchings per originated call. Then in the all-terrestrial system (N=1):

k(1) = 2 tandems x inter-regional traffic portion

+ 1 tandem x non-inter-regional traffic portion with 2 trunks in the connection

$$= 2 \times \frac{57}{70} + 1 \times \frac{8}{70} = 1.735 \tag{3.17}$$

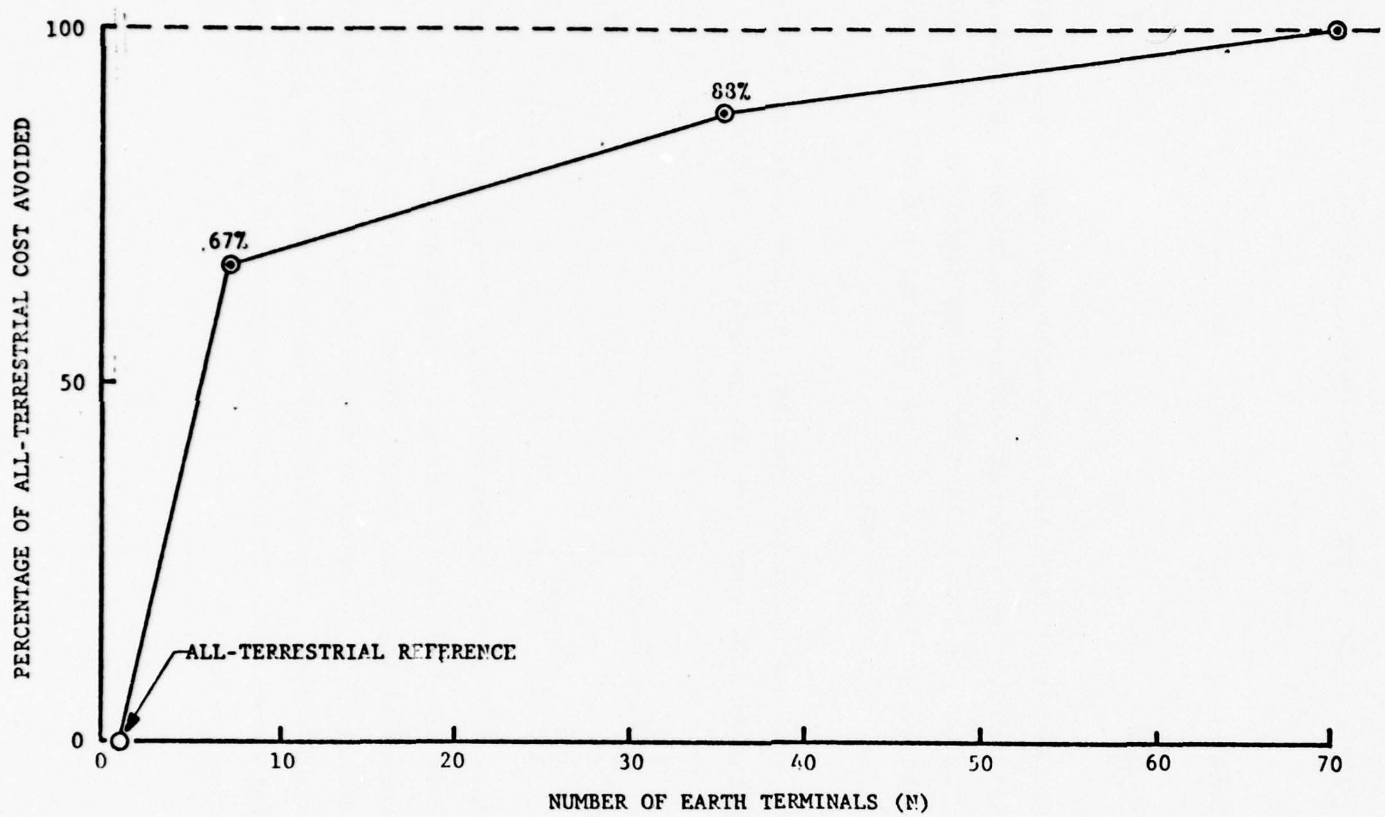


FIGURE 3.2.1 TERRESTRIAL TRANSMISSION COST SAVINGS AS A FUNCTION OF NUMBER OF EARTH TERMINALS

In the N=7 case,

$$k(7) = 1 \text{ tandem X non-inter-regional traffic}$$
$$= 1 \times \frac{8}{70} = 0.114. \qquad (3.18)$$

And in the N=35 case,

$$k(35) = 0. \qquad (3.19)$$

Again by assuming linear interpolation between cases considered above, we have obtained a functional relationship between the number of earth terminals and the cost savings for tandem switching. This relationship is plotted in Figure 3.2.2. At present, no dollar-cost data is known to apply to this relationship.

From the user traffic model described previously, we can deduce the switched traffic From/To matrix for any terminal area defined as:

$$M_i = \begin{bmatrix} E_{ii} & E_{i\text{-}out} \\ E_{i\text{-}m} & E_{i\text{-}tdm} \end{bmatrix} \qquad (3.20)$$

We have defined k(N) to be the average number of terrestrial tandem switchings over all calls for a N-terminal configuration, then the total tandem switched traffic in the entire network is k(N)E. We assume that the tandem traffic $E_{i\text{-}tdm}$ handled by terminal area i is proportional to its originating traffic. This assumption will be called the <u>proportional tandem traffic</u> assumption. Then from this assumption and the <u>balanced flow</u> assumption, we have

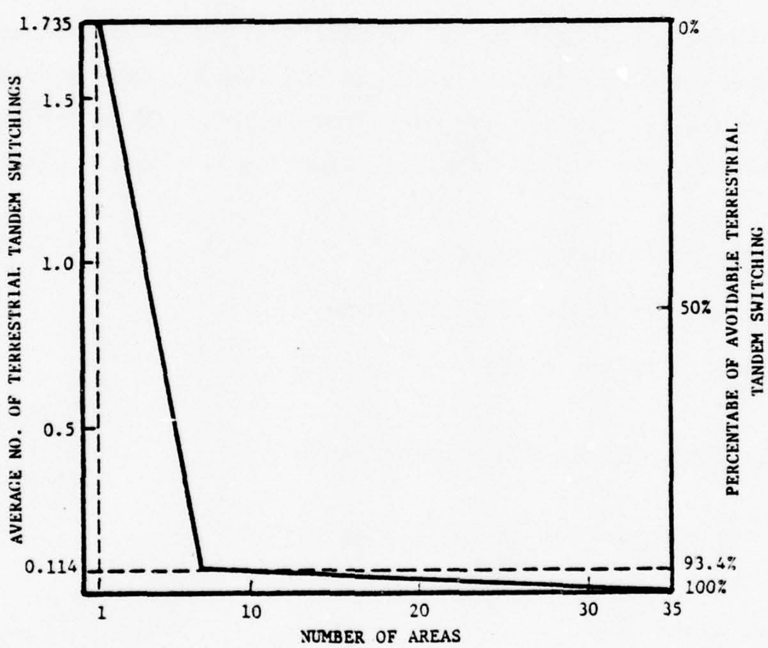


FIGURE 3.2.2 TERRESTRIAL TANDEM SWITCHING COST SAVING AS A FUNCTION OF NUMBER OF EARTH TERMINALS

$M_i =$

From: \ To:	Terminal Area	Other Areas
Terminal Area	$\frac{E_i^2}{E}$	$E_i(\frac{E-E_i}{E})$
Other Area	$E_i(\frac{E-E_i}{E})$	$k(N)E(\frac{E_i}{E})$

(3.21)

The model could be criticized on the grounds that if a terminal area is made small enough to include only a single PBX the E_{ii} term should be zero rather than E_i^2/E. To see how much "too large" E_i^2/E is for this situation, consider the "large location" from the DCS model. From (3.1),

Total Terminations = 1.8×10^3

Busy Hour Traffic = 0.1 Erl/Term

Inter-PBX Factor = 0.2

Thus, for this large single-PBX terminal area

$$E = (1.8 \times 10^3)\ (0.1)\ (0.2) = 36 \text{ Erl.}$$

Then the model gives

$$E_{ii} = \frac{(36)^2}{8 \times 10^3} = 0.162 \text{ Erl.},$$

which is only 0.45% of the total E_i of 36 Erlangs.

The case of most immediate interest is that of the baseline cost study where we assume all terminal areas support equal traffic, i.e.,

$$E_i = \frac{E}{N} \tag{3.22}$$

Then the From/To matrix becomes:

From: \ To:	Terminal Area	Other Areas
Terminal Area	$\frac{E}{N^2}$	$\frac{E}{N}\left(\frac{N-1}{N}\right)$
Other Areas	$\frac{E}{N}\left(\frac{N-1}{N}\right)$	$k(N)\ \frac{E}{N}$

(3.23)

Aggregated Total

The total switched traffic From/To matrix,

$$M = \sum_{1}^{N} M_i \tag{3.24}$$

with elements

$$M = \begin{bmatrix} E_{area} & E_{out} \\ E_{in} & E_{tdm} \end{bmatrix} \tag{3.25}$$

can be written out for the general case as follows:

$$M = \begin{array}{|c|c|} \hline \frac{1}{E}\sum_{1}^{N} E_i^2 & E - \frac{1}{E}\sum_{1}^{N} E_i^2 \\ \hline E - \frac{1}{E}\sum_{1}^{N} E_i^2 & k(N)\sum_{1}^{N} E_i \\ \hline \end{array}$$

In the equal traffic area case ($E_i = E/N$), M is as follows:

$$M = \begin{array}{|c|c|} \hline \frac{E}{N} & E\left(\frac{N-1}{N}\right) \\ \hline E\left(\frac{N-1}{N}\right) & k(N)E \\ \hline \end{array}$$

Note that in the large N limit

$$M = \begin{bmatrix} 0 & E \\ E & 0 \end{bmatrix} \tag{3.26}$$

The total originating traffic to be considered for satellite is just

$$\begin{aligned} E_{out} &= E - \frac{1}{E}\sum_{1}^{N} E_i^2, \text{ in general} \\ &= \frac{N-1}{N} E, \ E_i \text{ equal case,} \\ &= E, \text{ in the large N limit.} \end{aligned} \tag{3.27}$$

Thus far, we have only considered $N \leq 70$ in our terrestrial network model. For $N > 70$, it is envisioned that some of the 70 switches will be split into two or more smaller switches, each with its own satellite terminal, and the average user access line mileage will be reduced, while the total user access terminations will stay approximately the same. The average channel miles of user access line from [60] is 135 miles. This number will be reduced somewhat for $N > 70$, and the access savings model will be developed with assumptions from the satellite configuration in [60] in the next section.

3.3 OVERALL COST MODEL

The total system cost can be itemized as follows:

Total Cost = C(access) + C(digitizer) + C(activity detector) + C(terminal switching) + C(tandem switching) + C(terrestial transmission) + C(earth terminal) + C(satellite) (3.28)

For $N \leq 70$, C(access) and C(terminal switching) are not functions of terrestrial or satellite choice, and thus can be ignored here in the satellite cost savings. C(digitizer) and C(activity detector) are associated with voice encoding method and the choice of circuit switching versus packet switching. The C(tandem switching) and C(terrestrial transmission) are associated with terrestrial portions of the system, while C(earth terminal) and C(satellite) are associated with satellite portions of the system. Therefore, the incremental cost of a DAMA system can be written as:

IC(DAMA configuration) = C(digitizer) + C(activity detector),
if packet switching
− [C(tandem switching) + C(terrestrial transmission)]w/o DAMA
+ [C(tandem switching) + C(terrestrial transmission)]w DAMA
+ C(earth terminals) + C(satellite), $N \leq 70$. (3.29)

It is envisioned that almost all the future DCS CONUS network trunking will be leased from domestic commercial common carrier and hence there is no first cost involved and C(terrestrial transmission) will be C_t x (total trunk miles), where C_t is an unit annual cost per trunk mile. DCS switches will, on the other hand, be leased and dedicated to DCS use and, in general, be serving both terminating/originating and tandem traffic. The switching cost per switch will have a fixed cost in addition to the incremental cost per unit traffic.

Assuming that terminal switching is not a function of terrestrial or satellite choice and will always exist as a function of user traffic, then the terminal switching traffic and the switching fixed cost can be ignored in all candidate systems and C(tandem switching) is only a function of the incremental switching cost or C_s x (tandem traffic), where C_s is unit tandem switching cost in terms of annual cost.

With the above model, it is possible to simplify (3.29) so that the DAMA Incremental Cost (IC) is:

IC(N) = C(digitizer) x total number of users

+ C(activity detector) x total number of users, if DAMA employs packetized voice

- ΔC(tandem switching) - ΔC(terrestrial transmission)

+ C(earth terminal) + C(satellite)

= C(digitizer) x total number of users

+ C(activity detector) x total number of users, for packetized voice

$$- \sum_{1}^{N} C_s x\ k(N)E_i - C_t x[T(1) - T(N)]$$

+ C(earth terminals) + C(satellite) (3.30)

Using cost and access mileage figures from [60], the baseline terrestrial network cost model can be summarized as follows:

Total Cost / Cost Element	Monthly Cost per Mile	Annual Cost per Mile
Interswitch Trunk	$0.70	$ 8.5
Access Lines	$1.35	$16.2

Figure 3.3.1 shows the DAMA terrestrial channel-mile savings versus total number of satellite terminal areas.

At present, no realistic cost of switching per Erlang tandem traffic is known. To be conservative in estimating terrestrial cost savings this term will be taken as approximately zero compared to transmission costs. Thus, the transmission cost savings shown in Figure 3.3.1 will be used in system-wide cost tradeoffs.

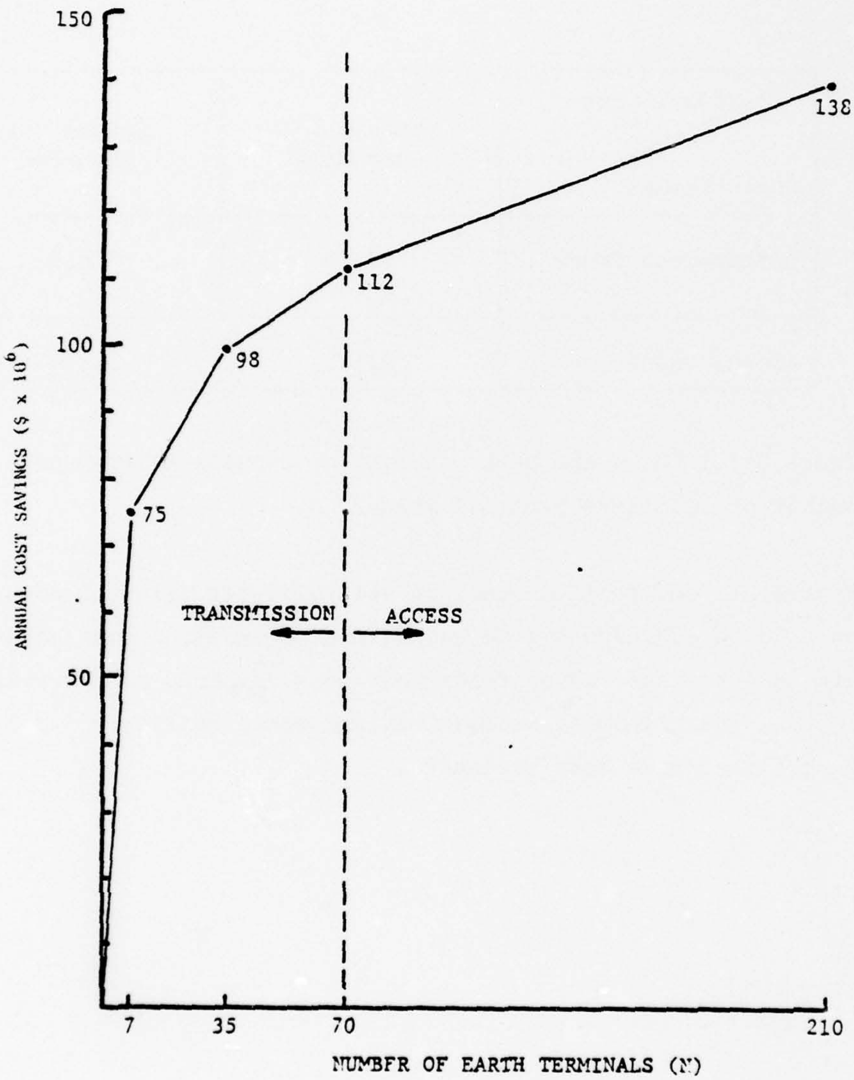


FIGURE 3.3.1 TOTAL TERRESTRIAL TRANSMISSION COST SAVINGS AS A FUNCTION OF NUMBER OF EARTH TERMINALS

4 DAMA CONCEPTS

4.1 CONCEPTUAL FRAMEWORK

4.1.1 Definition of Terms

The following definitions are provided to clarify and standardize the terminology used in this report.

Access: An access is the (temporary or permanent) use by an earth terminal of the satellite communication system.

Multiple Access (MA): MA is the access of the satellite communication system by a number of terminals over the same time period.

Demand Assignment Multiple Access (DAMA): DAMA refers to a multiple access system where the pool of users or potential accesses is larger than can be accommodated over the same time period - accesses being temporarily assigned on demand according to some protocol.

Mode-of-Access: The mode-of-access is the mode in which a user receives access to the satellite communication system. Some possible choices of mode-of-access are:

- Orthogonal channels (e.g., time, frequency, space)
- Non-orthoganal channels
- Non-orthogonal packet transmissions

Circuit-Switched: A DAMA system with orthogonal channels as the mode-of-access is said to be circuit-switched.

Packet-Switched: A DAMA system with a packet as the unit-of-access is said to be packet-switched.

Random Multiple Access (RMA): A RMA system is a DAMA system with a mode-of-access other than orthogonal channels. A contention packet-switched DAMA system such as ALOHA is an example of an RMA system.

4.1.2 Earth Terminal Interface Definition

The purpose of this section is to provide a definition of the interface between a demand access earth terminal and the common user voice/data network which connects to it. This definition is needed to clarify:

a. What performance characteristics are allocated to the candidate satellite access schemes; and

b. What cost elements are allocated to the earth terminal, space segment, and the terrestrial communication elements of the total system.

The definition may help reduce confusion in the above areas which arise because, by definition, any common-user network (e.g., an ordinary circuit-switched telephone network) constitutes in itself a DAMA system. The interswitch trunks and switch common equipment are accessed on demand and engineered on a statistical basis to serve a specified load at a specified grade of service; all users cannot be served simultaneously.

Earth Terminal Components

A satellite access earth terminal is assumed to consist of the basic functional blocks shown in Figure 4.1.1.

The key element in defining the common-user/earth-terminal interface is the block called the Digital Access Controller (DAC). The basic functions of the DAC are:

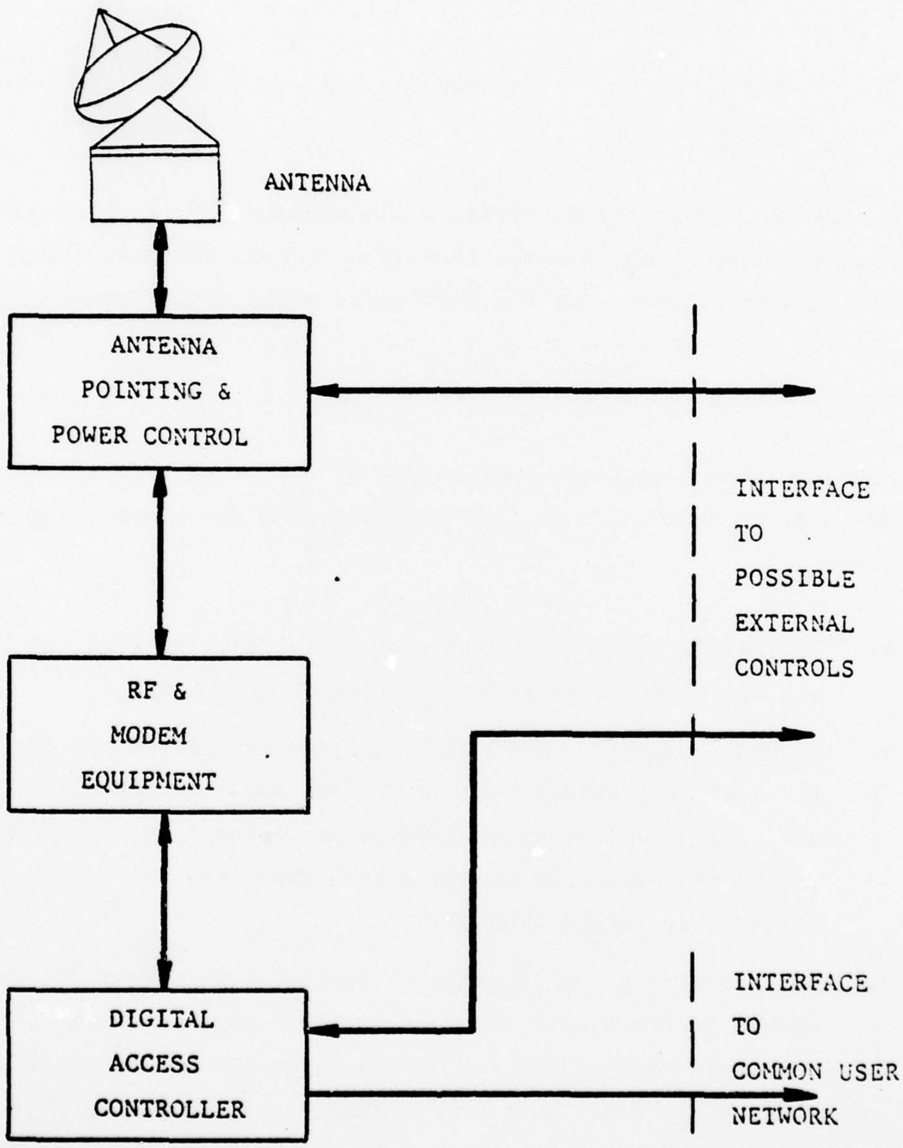


FIGURE 4.1.1 SATELLITE EARTH TERMINAL FUNCTIONAL COMPONENT

a. To buffer/multiplex/demultiplex between the satellite link data stream and the incoming/outgoing common-user network data streams, and

b. To carry out whatever protocols the particular DA/RA scheme requires.

The DAC is, in practical terms, a minicomputer of size, throughput and cost which depend on the data throughput rates, the complexity, and the buffering requirements of the particular DA/RA scheme used.

Further Interface Definition

With the above functional definition of the earth terminal we can now complete the definition of the interface with the common user network as follows:

a. Digital Interface: The information streams between the DAC and the common user network are in digital form.

b. Access Channels: Sufficient channels at appropriate speeds are assumed provided between the DAC and the terminal access area switch(es) so that blockage or degradation from contention within the satellite access system dominates any possible blockage in access channels.

c. Voice Digitization (Vocoder): Equipment is assumed to be placed in the access channels on the common user side of the interface when needed for analog voice A/D-D/A conversion. While it is possible - even likely - that voice digitization be placed physically at the DAC, the ET/network interface is conceptualized as digital rather than analog. The cost of this type of equipment is allocated to the earth terminal.

d. Common Channel Signaling: When the DAMA satellite system provides a circuit-switched-like facility, it is assumed that the interface presented to a common-user network switch is exactly like another switch. This is conceptually taken to be implemented by allocating an interfacing channel to digital common channel signaling, i.e., by simulating a CCIS interface.

e. Other Considerations: Other interface considerations such as error control, formats, etc. are not seen at this time to be pertinent to this study.

4.1.3 Categorization of DAMA Concepts

The number and variety of DAMA concepts which are applicable to a satellite communication system are becoming large. Several have already been invented by the authors in the course of this study and in preparation of the study proposal. To categorize and characterize these techniques requires attention to a number of issues. The increasing variety of techniques studied is increasing the list. The issues which are addressed in the following section (to varying degrees depending on applicability and understanding), as the techniques are described, are:

- Mode-of-access (circuit-switched, packet-switched, RMA, message-reservation, etc.)
- Measures of performance (blocking, delay, etc.)
- Satellite operation concept (RF transponder, processing, beam switching, etc.)
- Stability of the control protocol (runaway, lockup vulnerability, etc.)
- Centralized vs. distributed control structure
- Sensitivity to traffic model
- Performance/utilization tradeoff
- Maximum utilization (efficiency)
- Retry/retransmission policy
- Vulnerability of control to jamming
- Vulnerability/fairness response to user or terminal hogging

4.2 ACCESS SYSTEM PERFORMANCE

The communication performance of a DAMA system is measured fundamentally in two ways:

- Utilization of satellite channel capacity - which impacts cost, and
- Degradation of communication quality - which impacts satisfaction of user requirements.

The basic comparison between different access techniques is the tradeoff between the above two quantities.

The concept of satellite channel capacity, (i.e., maximum bit rate) is essentially independent of the type of DAMA technique under consideration; whereas the kind of communication quality degradation which occurs (with load) is qualitatively different for different access techniques.

Specifically, techniques which are circuit-switched degrade in terms of setup blocking rate and possibly delay. Data-oriented message and packet switching techniques degrade in terms of packet or message delay. A packetized-voice RMA system could degrade by random packet losses. In addition to pure load-induced degradations the introduction of priority and preemption operations creates additional degradations such as loss of a circuit-switched connection once established, or the creation of longer packet/message delays or losses in a packet or message-switched system.

Table 4.2.1 gives a comparison of modes of degradation for six categories of access and suitability of the match to voice and data requirements. The data characteristics are divided into "files" (long messages) and Q/R (query-response or short isolated messages or packets).

TABLE 4.2.1 PERFORMANCE DEGRADATION MODES FOR BASIC ACCESS CATEGORIES

Access Category	Modes of Degradation with Load	Immunity of Voice Quality to Load	Suitability For Data	
			Files	Q/R
(1) Preassigned (Not Switched)	Blocking	High	High	High
(2) Circuit Switched	Blocking, Setup Delay	High	High	Low
(3) Packet-SW W/Retrans	Random Delays	Low	Medium	High
(4) Packet-SW WO/Retrans	Packet Losses	Medium	Low	Low
(5) Packet-SW Limit, Retrans	Random Delays, Losses	Medium	Medium	Medium
(6) Data/Voice Mix of (3) & (4)	Packet Losses for Voice, Random Delays for Data	Medium	Medium	Medium

From this comparison we see that no single basic demand access technique (i.e., circuit vs. packet-switched) is well-matched in performance to both voice and data. Category 6 of the table is a synthetic hybrid which attempts to match both voice and data needs. A better synthesis would be one which sets up a non-degrading circuit (at the price of set-up delay/blocking) for voice calls and for data files/long messages, and uses random access packet switching (with retransmission) for short messages/packets.

4.2.1 Circuit-Switched DAMA Performance Measures

The circuit-switched DAMA techniques, as well as the fixed-assignment MA concept, all share blocking as the common mode of performance degradation with traffic load. A call set-up request for a fully occupied group of circuits is said to be blocked. What happens to blocked calls is the subject of numerous studies and models in the telecommunications literature and practice. For our comparative purposes we will use the commonly used and simplest model which assumes:

a. The population of voice terminals is large relative to the number of available circuits (infinite population).

b. Poisson arrivals and exponential holding times.

c. Blocked calls cleared (BCC).

From this model follows the so-called Erlang-B blocking formula:

$$B(n,a) = \frac{a^n}{n!} \Big/ \sum_{k=0}^{n} \frac{a^k}{k!} .$$

B(n,a) is the blocking probability; "n" is the number of circuits; and "a" is the traffic offered (in Erlangs). Subsequent subsections relate blocking through this Erlang-B formula to total traffic, total number of

channels, and number of earth terminals for the three basic circuit-switched concepts:

- Fixed Assignment (4.3.1)
- Directionally Variable Demand Assignment (4.3.2)
- Fully Variable Demand Assignment (4.3.3)

As a preliminary, we shall require the following results from Chapter 3. Let E be the total originating traffic, and assume:

- Balanced Flow
- Equal Terminal Area Traffic.

Then the traffic originating in any terminal area i and terminating in any area j is

$$E_{ij} = E/N^2.$$

Total outbound traffic originating in a traffic area or terminating inbound for a terminal area is

$$E_{i\text{-out}} = E_{i\text{-in}} = E(N-1)/N^2.$$

4.2.2 Packet-Switched DAMA Performance Measures

Packet-switching development has been motivated by the need to find switching methods which are effective for messages or data with holding times or lengths which are short with respect to circuit set-up times normally experienced with telephone circuit-switched equipment.

Virtually all packet-switching techniques (including RMA satellite techniques, e.g., ALOHA and derivatives) treated in the literature assume a requirement for all packets to "get through" - even if delayed.

This assumption leads to treating RMA as a queueing system. Performance for such systems is naturally measured by delay. (The mean or a percentile point of the delay distribution can be used as a single-parameter measure.) For these queueing system protocols, the fundamental tradeoff is delay vs. channel utilization or throughput. Their no-loss nature also leads to identity of offered load and throughput.

Subjective requirements for packetized voice real-time two-way communication leads to consideration of packet loss systems. In such systems, provided delay is bounded and acceptable, performance is measured by the packet loss rate (probability). For loss systems, offered load is not identified with equilibrium throughput, and the fundamental tradeoff is between offered load and packet loss rate. By measuring offered load as a fraction of channel capacity, offered load can be identified with the concept of utilization.

The following definitions will be utilized for packet-switched RMA satellite systems:

D = packet delay

S = throughput or carried load

U = channel utilization = offered load

G = aggregate contending channel traffic

(The above terms are all normalized with respect to bit rate capacity of a channel. S and U are also treated as probabilities.)

P_L = packet loss rate (probability)

q = single transmission packet loss probability

For equilibrium, the following relationships obtain:

$$q = 1 - S/G$$

$$S = U(1-P_L)$$

These relationships are general for both loss and lossless queueing systems, since for lossless systems $P_L = 0$ and $S = U$. In either case, a performance degradation measure (e.g., $E(D)$ or P_L) vs. channel utilization U describes the performance of the access system. The utilization U is related to required channel rate capacity C (burst rate) through the definition:

$$UC = \text{average information rate}$$

For a RMA packet-switched system (where each packet is handled individually) the average information rate is given for voice packets by:

$$\text{average information rate} = (b + H/t)\, d(2E)$$

where

E = aggregate originating traffic load (Erlangs)
b = digitizor bit rate
H = header bit count
t = packet time
d = voice activity duty factor

A typical duty factor is 50%. If a message (packet-string) switching concept is used (e.g., reservation or sub-slot capture), then the effective duty factor becomes 100%, since pauses within a long reserved string are not usable by other packets.

The relationship between U and G, while not a direct measure of performance, is helpful in facilitating insight into a protocol's performance characteristics, and is a basic step toward measuring performance.

4.3 CIRCUIT-SWITCHED DAMA CONCEPTS AND TECHNIQUES

Circuit-switched DAMA systems have the common characteristic that the unit of access to the satellite communication system is the circuit. Usually the circuit is a voice channel (in analog bandwidth or digital bit rate).

Channels to be assigned are provided by some orthogonal channelization of the available power and bandwidth. The principal choices of channelization are frequency (e.g., FDMA/FDM/FDMA/SCPC), time (e.g., TDMA), and code (CDMA/SSMA). For purposes of discussing the basic alternatives in circuit-swtched DAMA the particular channelization method is largely irrelevant. For implementation, however, the issue becomes important. Channelization techniques and their impact on choice of access technique are discussed in Chapter 5.

The following subsections define the basic choices of circuit-switched DAMA techniques and discuss existing systems as examples.

4.3.1 Fixed Assignment Multiple Access

Fixed assignment multiple-access (FAMA) refers to a multiple access satellite communication system where a fixed number of channels is permanently assigned between each pair of earth terminals. This concept is also known as "cables in the sky." It is, of course, not demand assignment at all, but is included here because it is a baseline multiple access technique against which possible advantages of DAMA techniques can be measured.

Fixed assignment is grouped along with circuit-switched DAMA techniques because it uses the circuit as the mode-of-access, and consequently shares blocking as the common mode of degradation.

The fixed assignment topology is illustrated in Figure 4.3.1. Assuming an equal number, L, of two-way channels (duplex circuits) between terminal pairs, then since there are N(N-1)/2 terminal pairs there are a total of $M = L \cdot N \cdot (N-1)$ one-way channels required of the satellite capacity. The number of channels between node pairs (L) is set by engineering to a given blocking level (grade-of-service).

Performance

Assuming calls can originate at either end of the channels, there is a total traffic of

$$E_{ij} + E_{ji} = 2E/N^2$$

between each pair. For L_{FA} two-way channels, the blocking is given by

$$B_{FA} = B(L_{FA}, 2E/N^2).$$

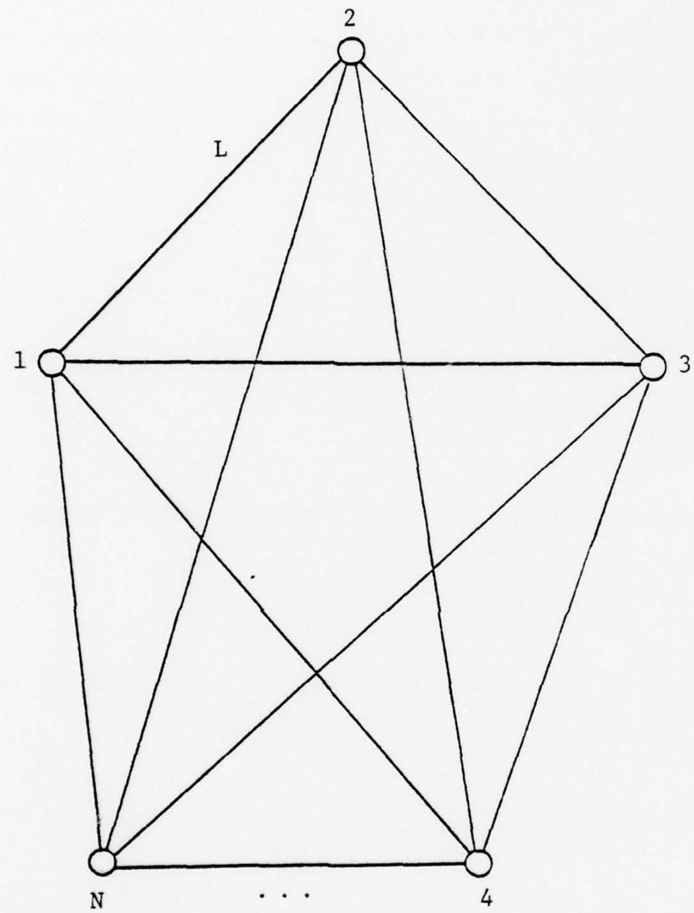
L_{FA} is sized to be the minimum such that B_{FA} is no larger than a specified grade-of-service, say 0.05.

4.3.2 Directionally Variable Demand Assignment

Directionally variable demand assignment (DVDA) subdivides a total set of M one-way channels provided by the satellite into N groups of L one-way channels, only one end of each channel being permanently assigned to an earth terminal.

$$M = L \cdot N.$$

This arrangement can be achieved for example by limiting each earth terminal to sending information over a unique set of L channels, and receiving on any of the remaining L(N-1) channels. This version is called destination variable demand assignment. Alternately, each



N = Terminals (Nodes)

L = Two-Way Channels/Node-Pair

FIGURE 4.3.1 AN EXAMPLE NETWORK WITH FIXED CHANNEL ASSIGNMENTS

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IMPLICATIONS OF DEMAND ASSIGNMENT FOR FUTURE SATELLITE COMMUNIC--ETC(U)
JUN 77 C G HILBORN, A C PAN, P J BOGERT DCA100-76-C-0060
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terminal could be arranged to receive on only L channels, and transmit on any of the remaining L(M-1) channels. In either case, one end of every one-way channel is permanently associated with one earth terminal, the other end must be assigned on demand by a setup protocol.

Blocking occurs for DVDA whenever all L sending (receiving) channels at an earth terminal are in use and a new call setup request arrives at that terminal or at another terminal and bound for the terminal with the fully occupied group. The L-channel group is sized to provide a given grade-of-service. Since the total set of M channels is splintered into N groups compared to N(N-1)/2 groups for fixed assignment, DVDA is a more efficient arrangement. The price of this increased efficiency is the complexity and delay associated with assigning one end of a channel for each new access.

Performance

The total traffic bidding for the L-channel group is

$$E_{i\text{-out}} + E_{i\text{-in}} = 2E(N-1)/N^2.$$

This is the total since both outgoing originations and incoming terminations require one receive channel. Then the blocking is given by

$$B_{DV} = B(L_{DV}, 2E(N-1)/N^2).$$

With minimum L_{DV} subject to making B_{DV} less than or equal to the chosen grade-of-service, the total number of (one-way) channels required is

$$M_{DV} = L_{DV}N.$$

4.3.3 Fully Variable Demand Assignment

Fully variable demand assignment (FVDA) utilizes demand assignment of a single pool of L (two-way) channels, with both ends of a channel assigned through an assignment protocol for the duration of a call. When the call terminates, the occupied channel reverts to the common pool. Since a single pool is used to carry the total traffic load, rather than N sub-pools as for directionally variable demand assignment, fully variable demand assignment is the most efficient circuit-demand assignment technique. The price for this increased efficiency in utilization of satellite capacity is in possible implementation complexity/cost: every terminal must be able to receive and send on any and all channels.

Blocking occurs for FVDA whenever a new call setup arrives while the entire pool of L two-way channels is busy. The L-channel group is sized to provide a given grade-of-service for the total traffic load. Significant advantage of FVDA over DVDA for the DCS environment is that the performance is sensitive only to the total traffic and not affected by shifts within the To/From traffic matrix.

Performance

For fully variable demand assignment, a single pool of L_{FV} (two-way) channels is available for assignment to any call between any pair of terminal areas. The total inter-terminal area originating traffic is

$$N\, E_{i\text{-out}} = e(N-1)/N.$$

Thus, L_{FV} is the minimum such that

$$B_{FV} = B(L_{FV},\ E(N-1)/N)$$

is less than or equal to the required grade-of-service, and the total number of one-way channels which the satellite capacity must supply is

$$M_{FV} = 2\ L_{FV}.$$

At the large total traffic levels of this study (2000-8000 Erl.) the channel group is essentially 100% efficient. That is, required size L_{FV} is approximately equal to the carried load:

$$L_{FV} \cong (1-B)\ \frac{E(N-1)}{N}$$

$$= 0.95\ \frac{E(N-1)}{N}\ ,$$

for an allowed blocking B = 0.05.

In addition to blocking, any implementation of FVDA will require a finite time to allocate the circuit required for the call. This time should be little more than two round trips (0.56 seconds) for a centralized request or polled implementation, or little more than one round trip for a distributed contention system (e.g., SPADE) with the penalty of possible retrials required after unsuccessful bids.

4.3.4 SPADE [73]

The SPADE system (SPADE is an acronym denoting "Single channel per carrier, PCM, multiple-Access Demand-assignment Equipment") provides an existing example of a fully variable demand assignment (FVDA) satellite communication system. SPADE was developed by COMSAT Laboratories on behalf of INTELSAT.

Channelization for SPADE is effected by single channel per carrier FDMA. Each carrier is 4-phase modulated with 7-bit PCM digitized voice. The system is designed to provide up to 397 two-way voice channels, plus

a common signaling channel (CSC). The frequency allocation plan utilizes 45 kHz spacing of 794 one-way voice channels, and a 160 kHz bandwidth for the CSC - for a total bandwidth of 36 MHz - corresponding to one INTELSAT IV transponder. Operation with that transponder at the design bit error rate (10^{-4}) requires an earth terminal G/T of 40.7 dBW/°K.

The demand assignment protocol for SPADE is fully distributed, i.e., there is no central control and every ET uses the same algorithm. Each ET keeps a table of busy and idle channels by monitoring the CSC. The CSC is a TDM channel with one burst slot permanently assigned to each of up to 49 earth terminals. To seize a channel an ET chooses at random an idle channel and broadcasts that intent over the CSC. Simultaneous seizures (within the same round trip delay) are resolved in favor of the first to arrive. Successful seizures thus take one round trip time (0.28 sec.) plus at most a CSC-TDM frame time (50 ms), or 0.33 sec. Similarly, a channel is released in about 0.33 sec. by an announcement over the CSC. Assuming an average holding time for voice calls of 3 minutes or more, the DAMA overhead time to allocate and deallocate a channel is no more than about 0.4% of call holding time.

4.3.5 MAT-1 [52]

MAT-1 is a TDMA technology-based destination variable demand assignment experimental system developed by COMSAT Laboratories.

The technique divides a 50 Mbps satellite channel into short time slots or bursts, and a frame of time slots is established. Each earth station is assigned a position in this frame based on time of entry into the network. During its burst time each station transmits a fixed number of control bits followed by a variable number of data bits. The bursts are separated by a short guard time. One of the earth stations acts as a reference to determine the start of each burst frame.

A single channel in this system consists of 8 bits of data occurring once each frame (125 μs in this case). If the burst had a fixed duration, then each channel would have a fixed place in the frame, and would be permanently assigned on the transmitter end. In the MAT-1 system, the frame fraction allocated to each earth terminal is variable and can be added or subtracted from the frame to follow traffic fluxuations. Channel assignment or de-assignment requires about 0.3 sec.

A control word is transmitted when a channel between two earth stations is set up, and another control word is transmitted when the channel is "torn down". Control bits are also transmitted to indicate the number of allocated channels actually in use so that slack channels can be reallocated and burst durations varied according to some "fairness" criterion. Each station monitors the control bits of all other stations to determine which channels should be demultiplexed and to determine its proper burst length (number of allocated channels).and time of burst. Thus, the control channel is distributed through the frame, and the control system is distributed through the network.

The MAT-1 implementation of DVDA yields about 700 8-bit PCM voice channels (64 kbps/channel) for a total data throughput of 44.8 Mbps. The overall channel utilization efficiency is then calculated as:

$$\frac{44.8}{50} = 0.90$$

The similar calculation for SPADE in (bandlimited) utilization of the 36 MHz channel gives:

$$\frac{800 \times 0.056}{36 \times 2/1.2} = 0.75$$

This difference is not related to the demand assignment technique; it is related solely to the respective bandwidth and timing overhead and margin allowed in the FDMA versus TDMA implementations.

4.3.6 Comparison of Circuit-Switched DAMA Performance

The three basic circuit-switched DAMA choices, fixed assignment, directionally variable demand assignment, and fully variable demand assignment, can now be compared in terms of satellite capacity required to support the same traffic at a fixed grade-of-service. Using the implicit relationships for the total number of (one-way) voice channels which the satellite capacity must support (developed in the preceding sections), the plot shown in Figure 4.3.2 was generated. The discontinuous nature of the FAMA and DVDA functions is due to the number of per terminal-pair and per terminal channels being a constant integer over a range of N. Each step down is for one fewer channel per ET or ET pair. The small-N droop is due to the (N-1)/N factor between total originating traffic E and total inter-area traffic.

For this plot a total originating traffic of 8000 Erlangs was used with a grade-of-service of 5% blocking.

Set-up delays on the order of 1/2 second are quite acceptable for circuit-switched voice calls with holding times on the order of 2-6 minutes: The unavailable capacity during set-up is negligible and the waiting time is presumably acceptable to a common-user community.

On the other hand, for Q/R and isolated small packet data, such delays may be unacceptable. From a user-satisfaction viewpoint, such delays present a problem. From an efficiency viewpoint, such delays are

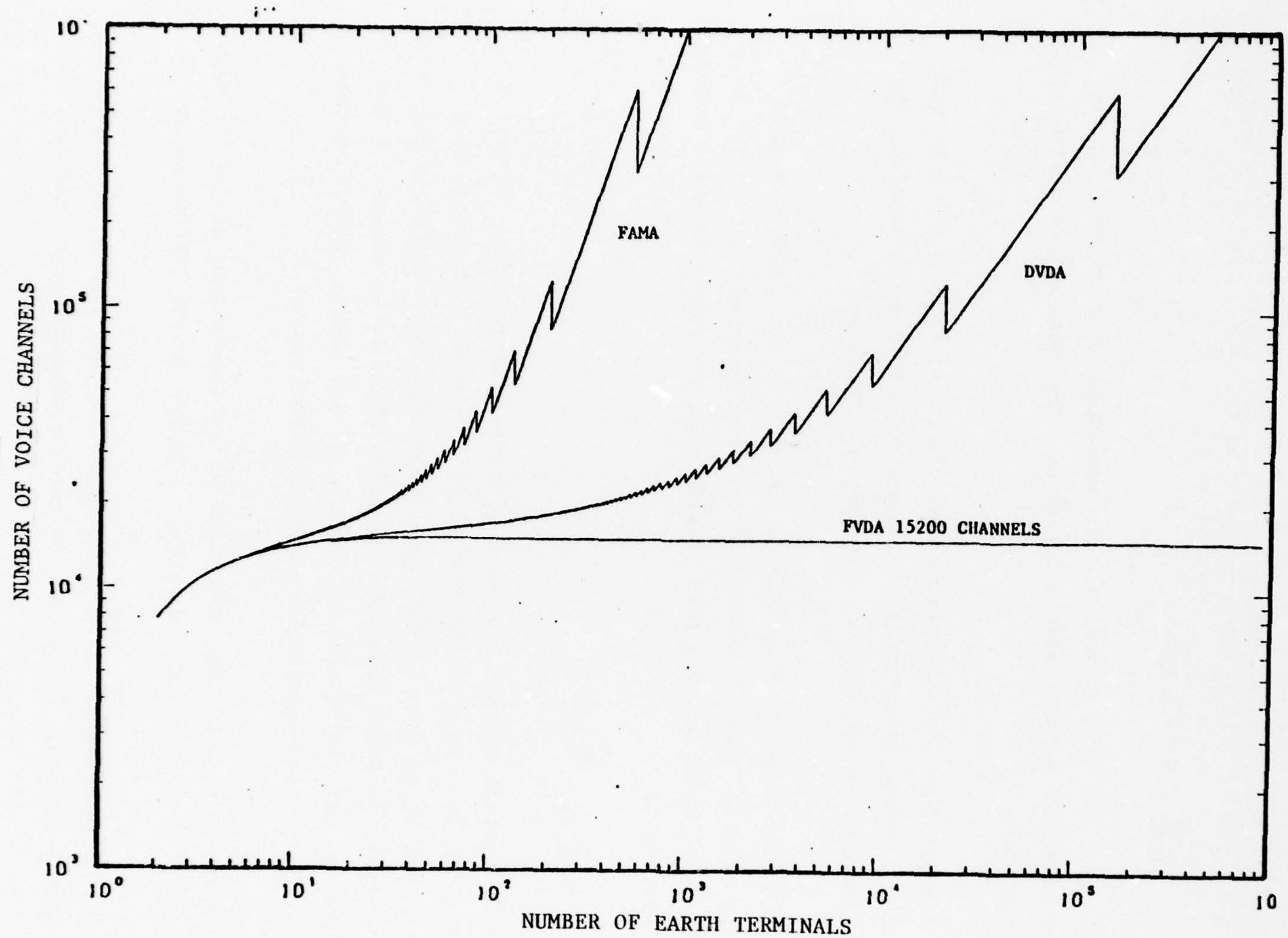


FIGURE 4.3.2 VOICE CHANNEL REQUIREMENTS FOR CIRCUIT-SWITCHED DAMA ALTERNATIVES

severe. For example, for a 1000 bit packet transmitted on a 16 kbps channel, the total packet time is

$$\frac{1000}{16000} = 0.0625 \text{ sec.}$$

Assuming a total time to allocate and deallocate the channel for the single packet of about 1 second the channel is being utilized with only 6% efficiency.

A possibility to consider is that the proportion of short holding time data traffic is so small that even if ineffeciently handled, the total increase in required capacity may not be large.

Consider holding times (packet times) of x sec. The effective holding time including a set-up and takedown delay of y is x + y. If e is the total short holding time traffic then the effective short holding time traffic is

$$E_{eff} = \frac{x + y}{x} e.$$

For $y \simeq 1$ second and $x = 0.05$ to 0.1 second, the magnifier $(x + y)/x$ is about 10 to 20. Thus to add even 1% traffic with 0.05 second holding time to the total long holding time (voice and file) traffic requires a 20% increase in total satellite capacity. This is an example of the fundamental motivations for considering random access, packet-switched systems in a mixed data/voice common-user environment.

Another issue of the comparison is the sensitivity of the technique to <u>traffic shifts</u> or, equivalently, to errors in the traffic model.

The FAMA approach requires the most detailed traffic knowledge, with the assignments being made to match a terminal pair traffic total:

$$E(i,j \text{ pair}) = E_{ij} + E_{ji}$$

Thus any shifts or modeling inaccuracy in traffic at this level will cause imbalance and higher blocking between the terminal pairs than designed. Some user pairs may get poorer service than nominal and others may receive better service than nominal.

DVDA is less sensitive since it operates at a higher level of traffic aggregation. Blocking is determined on a total per earth terminal traffic basis:

$$E(i\text{-sat}) = E_{i\text{-out}} + E_{i\text{-in}} = \sum_{\substack{j=1 \\ j \neq i}}^{N} E_{ij} + \sum_{\substack{i=1 \\ i \neq j}}^{N} E_{ij}$$

The MA[illegible]sign of a variable frame fraction allows tracking these shifts.

FVDA is the least sensitive to traffic shifts or modeling errors since it operates (and blocks) only on the total inter-area traffic:

$$E(\text{sat}) = \sum_{i \neq j} E_{ij} = E - \sum_{i=1}^{N} E_{ii}.$$

A subtle design issue, however, is that even when a fully-variable assignment algorithm is used, a terminal will operate at an average up-link power determined by E(i-sat), which for the equal traffic case is:

$$\frac{E(i\text{-sat})}{2} = \frac{E(N-1)}{N^2} \simeq \frac{E}{N}$$

If the power amplifier is sized for this load, the shifts allowed by the FVDA algorithm cannot be accommodated. Similarly, in a SPADE-type (SCPC) implementation there are per channel cost items per terminal which pushes toward only equipping for about E/N traffic per terminal. If this compromise is made, the system is no longer true FVDA. Yet some compromises along these lines should be investigated. For example, a factor-of-two over E/N channel equippage and P.A. average power design will allow for moderate shifts in traffic without performance penalty.

4.4 PACKET-SWITCHED DAMA CONCEPTS

In 1970, Norman Abramson [1] of the University of Hawaii proposed a random multiplexing technique, termed ALOHA, to permit a large number of stations to communication over a single ground radio channel. The ALOHA concept has subsequently been considered as a random multiple access technique for a satellite channel [4] and has spawned a large number of studies which attempt to understand the performance under various circumstances, and moreover has been followed by a large number of proposals for similar techniques which attempt to overcome the limitations of the original technique and subsequent techniques. In describing these techniques we are intending only to present a brief summary of the protocols and analysis, as well as suggest some new approaches based on voice requirements, and overcoming the instability and low efficiency of ALOHA.

Although other versions have been studied, all references in the following to "packets" will be for fixed-length packets. Longer messages consist of several packets.

4.4.1 Pure ALOHA

Protocol

In a pure ALOHA protocol, each earth terminal (ET) transmits packets of fixed length. The packets are transmitted according to some random, but mutually independent (among ETs) rule in the common satellite channel as illustrated in Figure 4.4.1. The inter-ET transmissions are conducted among the ETs on a contention basis without central coordination or control. In the event that packets from different ETs overlap in time, by any amount, errors are created. These destroyed packets are detected and retransmitted by their respective ETs.

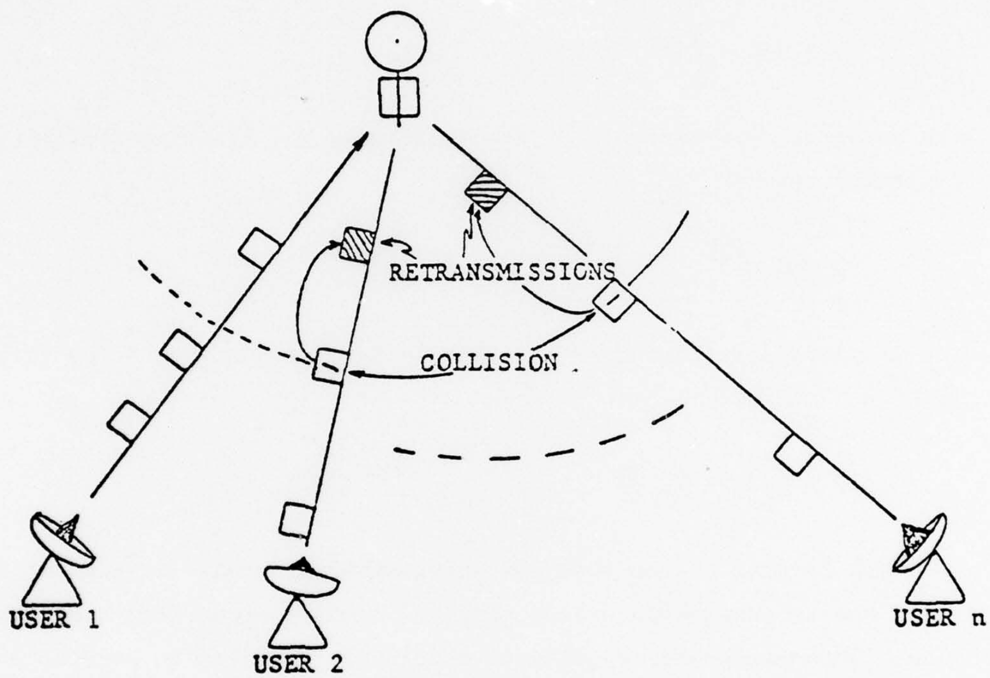


FIGURE 4.4.1 ALOHA SYSTEM CONCEPT

Performance

For pure ALOHA, the relationship between the normalized utilization, U, and the normalized channel rate, G, for a finite number of equal earth terminals is given by:

$$U = G(1 - 2G/N)^{N-1},$$

which converges very rapidly to the well-known [1] limiting (infinite population) result

$$U = G\, e^{-2G},$$

which is plotted in Figure 4.4.2. The maximum utilization $\hat{U}$ (usually called the "capacity") is

$$\hat{U} = 1/2e = 0.184$$

These results assume that the retransmission times are randomized so that the retransmissions plus original transmissions form a Poisson stream. Some randomization time is critically required to prevent a deadlock cycle between two or more terminals whose packets have collided.

While the distributed control nature of ALOHA is highly attractive, it suffers from low efficiency (less than 18%) and an instability evident in the <u>two</u> values of channel traffic (G) for each value of throughput (U).

Delay for satellite ALOHA-switched packets is from three sources:

- Propagation time (0.27 sec. x number of trips)
- Queueing time
- Retransmission time randomization

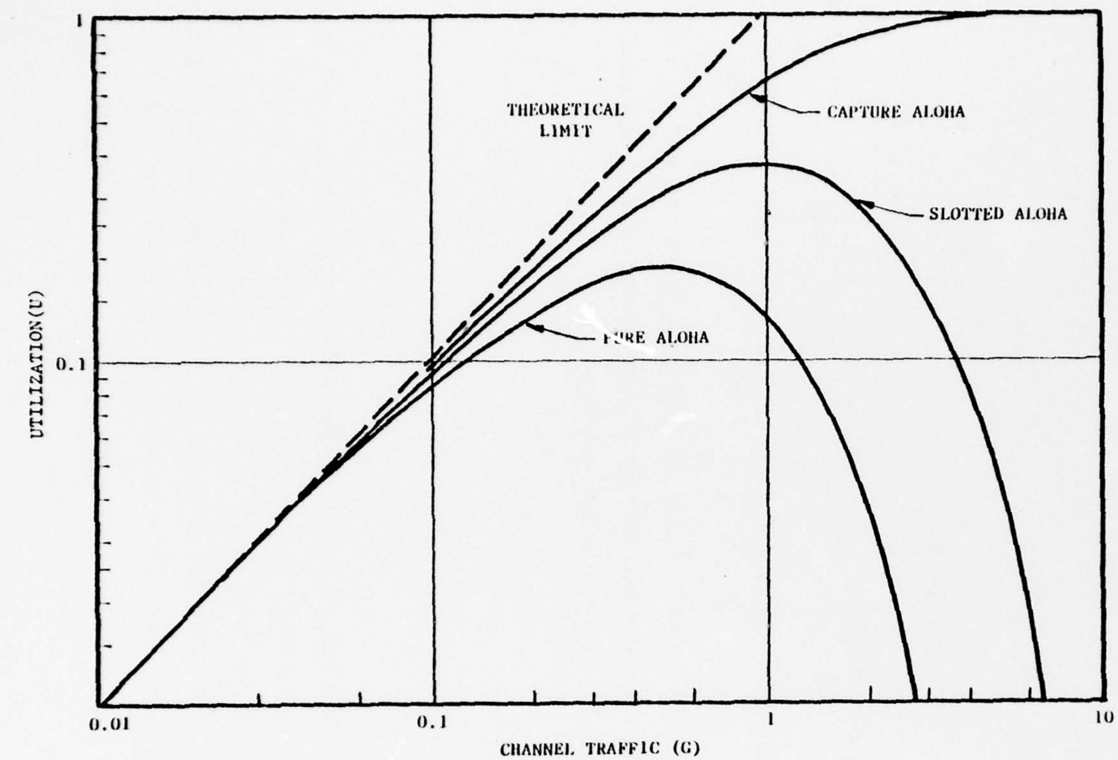


FIGURE 4.4.2 UTILIZATION AS A FUNCTION OF OFFERED LOAD: PURE, SLOTTED, AND CAPTURE ALOHA

For large N, the average queueing component is small compared to one-trip propagation time, and for the large capacity speech packet system contemplated (packet burst times on the order of 1 microsecond) randomization time is small compared to 0.27 sec. Hence, mean delay is given by one trip time T = 0.27 sec. times the average number of transmissions/packet [28]:

$$\bar{D} \cong T\ G/U = T\ e^{2G}.$$

In terms of utilization U,

$$U = \frac{1}{2}\,\frac{\ln(\bar{D}/T)}{\bar{D}/T}$$

which is plotted in Figure 4.4.3.

All of the packets which are delayed more than one or two retransmissions are unusable for two-way voice and serve only to clutter the channel with instability-producing traffic.

4.4.2 Slotted ALOHA

Protocol

The low utilization factor of pure ALOHA is due to the completely random emissions of packets from each user. A method of improving the channel utilization by "slotting" time into segments whose duration is exactly equal to the transmission time of a single packet was proposed by Kleinrock and Lam [35]. This concept is called slotted ALOHA.

Performance

The channel utilization, U, and the channel traffic, G, can be shown as related by (see Figure 4.4.2):

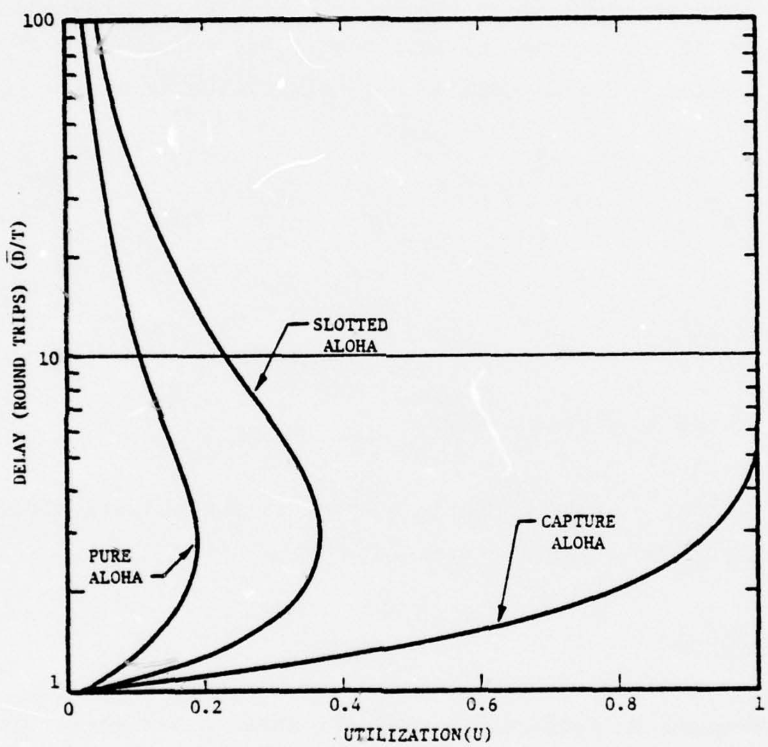


FIGURE 4.4.3 NORMALIZED MEAN DELAY AS A FUNCTION OF UTILIZATION FOR PURE, SLOTTED, AND CAPTURE ALOHA

$$U = G(1 - G/N)^{N-1} \rightarrow G\, e^{-G}$$

which results in a maximum channel utilization of

$$\hat{U} = 1/e = 0.368$$

Thus, slotted ALOHA achieves twice the maximum channel utilization of a pure ALOHA system at the expense of introducing the requirement of time frame synchronization. The mean delay vs. utilization is similarly improved to [24]:

$$\overline{D} = T\, e^{G},$$

or

$$U = \frac{\ln(\overline{D}/T)}{\overline{D}/T}$$

which is illustrated in Figure 4.4.3.

Like pure ALOHA, slotted ALOHA is subject to instability without imposition of extra control structure.

4.4.3 CAPTURE ALOHA

Some improvement in performance over the equal-power/equal-traffic cases for pure and slotted ALOHA have been recognized. In particular, Roberts [55] has pointed out that for a given geographic distribution of equal-power earth terminals, some terminals will be closer to the satellite than others and will enjoy an RF "capture" effect. The capture effect occurs whenever one signal is about 6 dB or more stronger than another. The stronger signal packet survives the collision with low error probability and the weaker signal packet is lost. Similarly, it has been proposed to give some ETs more power than others to increase capture effects [47a]. The capture effect clearly increases throughput since collisions are not all destructive.

Protocol

Hilborn [28] has proposed a processing satellite which creates a perfect capture effect. This CAPTURE concept operates as follows (see Figure 4.4.4): Each earth station transmits on a separate (orthogonal) UP channel of burst rate (capacity) C equal to the single DOWN channel. When one or more packets is received on the UP channels, the onboard controller selects one packet (when two or more contend) for transmission on the DOWN channel. Thus, one packet is always successful even in multipacket contentions.

The onboard selection algorithm could be chosen on a number of possible bases such as:

- Random selection
- Priority selection

If the onboard selection is done at IF (without demodulation), the only priority possible is per transmitting earth terminal (which priority could be re-orderable by ground control). If the onboard selection is done after demodulation at baseband then priority selection is possible on a per packet basis, by the simple inclusion of a few priority bits in the packet. Moreover, the onboard demod/remod process gives about 4 dB power advantage over conventional transponders [14,42].

The most significant advantage of this capture technique over transponder ALOHA, however, is that without adding any new control protocol for earth terminals throughput is increased and instability is eliminated. The throughput/channel traffic relationship for CAPTURE ALOHA is monotone (stable):

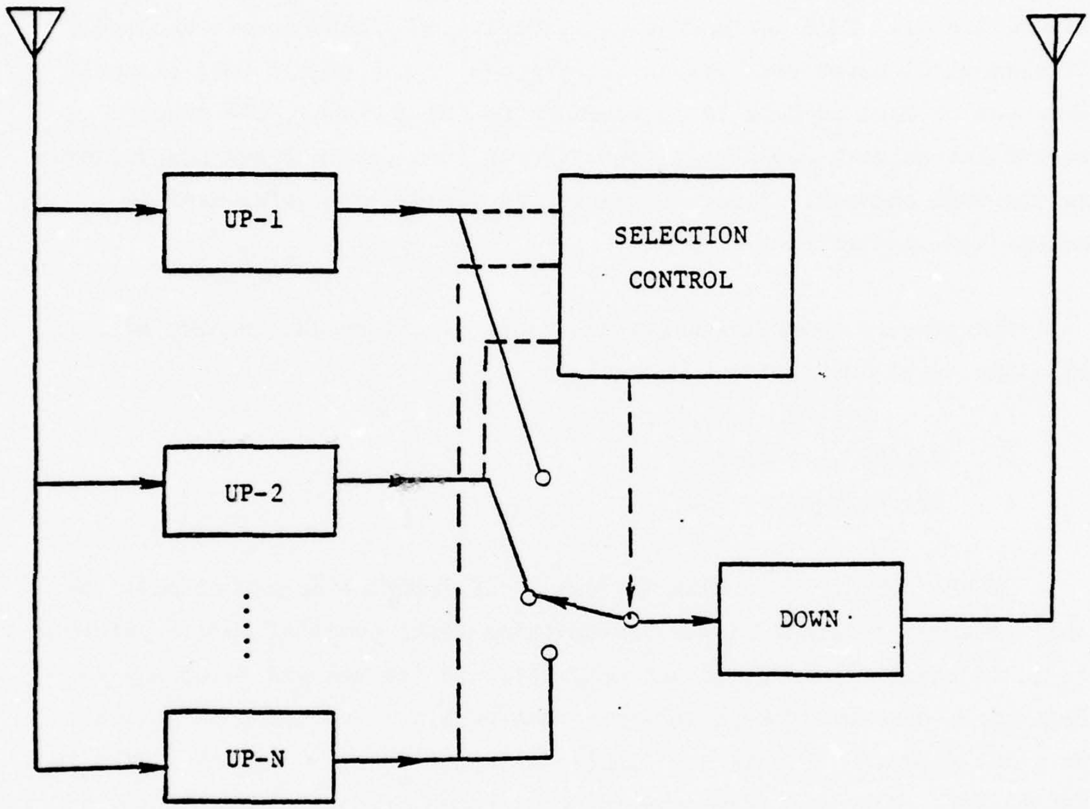


FIGURE 4.4.4 CAPTURE PROCESSING SATELLITE

$$U = 1 - (1 - G/N)^N \rightarrow 1 - e^{-G}.$$

$$\hat{U} = \text{MAX } U = 1$$

The limiting G vs. U is plotted in Figure 4.4.2 along with pure and slotted ALOHA. The mean delay is accordingly given by

$$\bar{D} = T\, G/(1 - e^{-G})$$

$$= \frac{\ln[1/1-U)]}{U} T,$$

which is plotted in Figure 4.4.3.

Possible disadvantages of CAPTURE ALOHA are:

1. It requires a new processing satellite of higher technology risk and cost than established RF transponder techniques.
2. The requirement for orthogonal UP-link channels uses up spectrum rapidly and increases with N, tending to limit the number of ETs.

The first objection to cost is more than offset by the gains in efficiency of the technique and in power by using a demodulating satellite, both of which lower ET cost. The second objection can be partially offset by using spectrum-conserving orthogonalization techniques such as:

- Spot beams for frequency reuse
- Cross polarization for frequency reuse
- Higher bits/symbol modulation (e.g., 16-ϕ PSK)

Finally, we remark that all of the spectrum-spreading of the total UP-link has a direct A-J advantage.

4.4.4 Other ALOHA Variations

A number of other additions and variations on the basic ALOHA protocol have been proposed. These are discussed in the following paragraphs.

4.4.4.1 Overlap ALOHA

In pure ALOHA the basic throughput vs. channel traffic and delay relationships assume that two packets with any amount of overlap in time are unusable and must be retransmitted. Preamble and address bits are critical to correct packet handling but some errors in speech packets may be tolerable. If a fraction of overlap Z ($0 \leq Z \leq 1$) is permitted, then the basic pure ALOHA G vs. U relationship becomes

$$U = G\ e^{-G(2-Z)}$$

Thus, tolerating some overlap permits performance between pure ALOHA and slotted ALOHA.

4.4.4.2 Carrier-Sense Multiple Access

In Carrier-Sense Multiple Access (CSMA) attempts to avoid wasting channel capacity with destructive collisions of pure ALOHA by having each terminal listen for the presence of carrier due to another terminal's transmission and refraining from immediate transmission, and using some rescheduling rule [39,40,69]. Three different rescheduling rules have been proposed. These rules are called 1-persistent, non-persistent, and p-persistent CSMA.

This class of protocols was originally proposed for ground radio channels where propagation delay is short compared to packet time. For

a high-capacity voice packet satellite system, the opposite situation obtains: the packet time is very short compared to propagation delay, and this class of protocols has no advantage over pure ALOHA.

4.4.5 Reservations

The previous discussion of ALOHA-type RMA techniques are most applicable to traffic composed of one-packet messages. A number of variations called reservation schemes have been proposed to increase throughput for multiple-packet messages. These schemes have in common the characteristic of increasing the minimum delay and increasing maximum throughput. In a sense, they are a compromise between pure packet-switching and pure circuit-switching. Because the relatively large minimum delay and the multi-packet string model both seem poorly matched to voice requirements, these techniques are not being emphasized in this study. The following two well-known techniques are included for completeness.

4.4.5.1 Robert's Reservation

Under this protocol [57], the channel is divided into two (time frame) subchannels. One subchannel, operated in slotted ALOHA, is for reservation requests. The other subchannel, operated in dedicated mode, is for multi-packet data. Because of the broadcast nature of the channel, all users can hear the successful requests for data slots and appropriately schedule their own data transmissions (if requesting) and requests. Since the request channel is operated in slotted ALOHA mode, there will be request collisions and instability problems.

While the reservations introduce additional elements of control over simple ALOHA, Robert's Reservation distributes control responsibility to all ETs.

4.4.5.2 Round-Robin Reservation [8]

The Round-Robin (RR) technique uses a slotted channel with a basic TDMA frame structure. Each of N ETs is assigned a one-packet slot per N-slot frame. Thus, the underlying structure is similar to a single-channel-per-ET directionally variable demand assignment. In addition, an active ET can dynamically acquire use of the slots assigned to other ETs by inserting a reservation request into the preamble of packets it is transmitting in its own slot. A previously inactive ET whose slot has been acquired by another ET can reclaim its slot by sending at will in its own slot and deliberately creating a conflict. The mutual protocol requires cessation of use of a slot by the non-owning terminal.

The RR technique is thus similar to CSMA with the voice-packet time shortness extended to a frame time. Nevertheless, the frame time is also short relative to a round trip time.

4.4.6 Finite Retransmission (Loss) Protocols

Since for (two-way, real-time) voice packets, packets delayed by more than one or two retransmission are useless and serve only to clutter an ALOHA-type channel with destabilizing retransmissions, we consider a new class of ALOHA-type protocols which either use no retransmission of unsuccessful packets, or at most a few retrys. A significant result of this approach is that the ALOHA protocol is thereby stabilized.

For packet-loss systems operating with digitized voice packets, the packet loss rate (probability) P_L is the appropriate measure of performance degradation with load. Preliminary findings are that a P_L in the range of 5% to 15% is tolerable for short (20 ms) speech packets (see Chapter 6).

We first consider modifying the basic protocol for pure and slotted ALOHA, and CAPTURE by allowing no retransmissions, then generalize to allow a finite number, n, retrys.

4.4.6.1 No Retransmissions

From Section 4.2.2 the probability of single-attempt packet loss is given by:

$$q = 1 - S/G.$$

For a policy of no retransmissions

$$P_L = q$$

The dual interpretation of U as either offered load or channel utilization is again emphasized. U is not throughput, S, since

$$S = (1 - P_L)U.$$

Finally, note that from the above, U = G.

The utilization vs. channel load relationships developed for pure and slotted ALOHA and CAPTURE in Sections 4.4.1 - 4.4.3 can now be directly applied to yield the packet loss performance for these RMA techniques:

<u>PURE ALOHA</u>

$$P_L = 1 - (1 - 2G/N)^{N-1} \rightarrow 1 - e^{-2G}$$

SLOTTED ALOHA

$$P_L = 1 - (1 - G/N)^{N-1} \rightarrow 1 - e^{-G}$$

CAPTURE ALOHA

$$P_L = 1 - \frac{1}{G} [1 - (1 - G/N)^N] \rightarrow 1 - \frac{(1-e^{-G})}{G}$$

Note also that for no retransmission these relationships can be made explicit in U by U = G. These packet loss performances are plotted in Figure 4.4.5, in the $N\rightarrow\infty$ limiting case. Table 4.4.1 compares the offered load or utilization for the above three cases at 5, 10, and 15% P_L, and the resulting satellite burst rate (capacity) requirements to support an aggregate traffic load of 8000 Erlangs of voice traffic at 4 and 16 kbps digitization rates.

Finally we remark that since $G = U$ and P_L vs. U *is single-valued*, there is no instability problem. The instability of the pure and slotted ALOHA is a result of retransmissions.

4.4.6.2 Finite Retransmissions

We now generalize the previous analysis of no-retransmission to permit a finite number of retrials, n. As for the previous queueing ALOHA analysis, retransmission times are assumed to be randomized sufficiently to consider the retransmissions to form a Poisson stream for pure ALOHA, and to form an independent sequence for slotted ALOHA and CAPTURE.

If q is the single-transmission packet loss probability, overall loss, which is the event measured by P_L occurs when n+1 single-attempt losses occur:

$$P_L = q^{n+1}$$

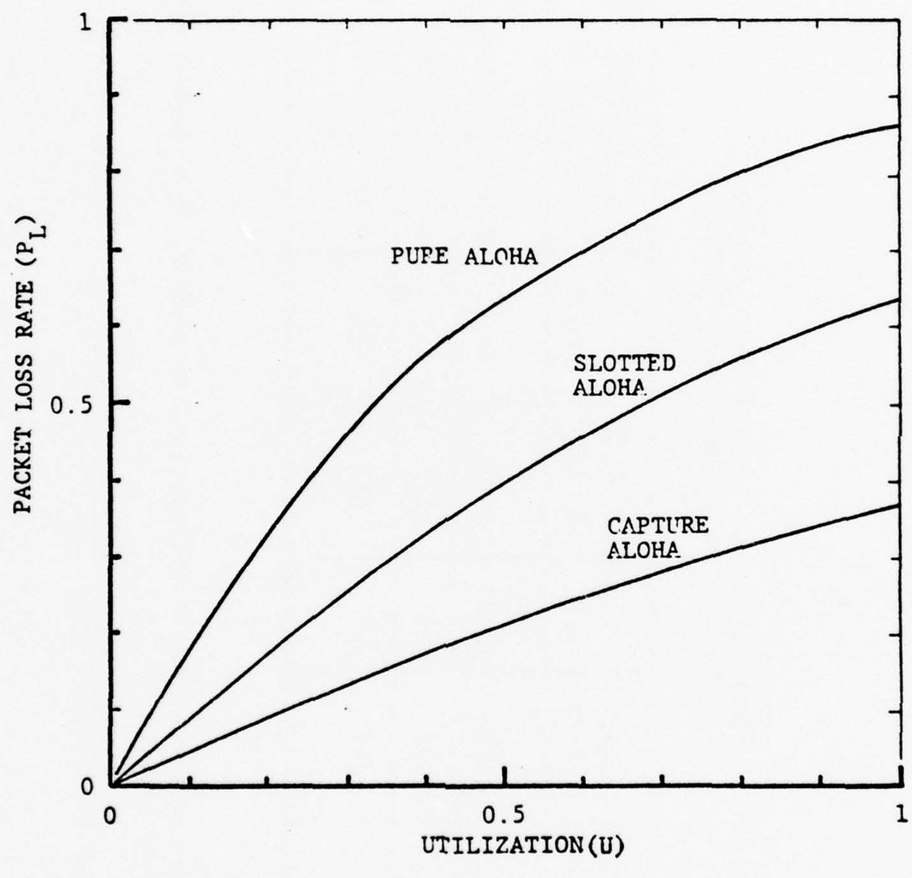


FIGURE 4.4.5 PACKET LOSS RATE FOR PURE, SLOTTED, AND CAPTURE ALOHA AS A FUNCTION OF OFFERED LOAD WITH A 0-RETRANSMISSION POLICY

TABLE 4.4.1 EFFICIENCY AND REQUIRED CAPACITY FOR ALOHA DERIVATIVE PACKETIZED VOICE RMA

P_L	PURE ALOHA			SLOTTED ALOHA			CAPTURE ALOHA		
	EFF.	16 KBPS	4 KBPS	EFF.	16 KBPS	4 KBPS	EFF.	16 KBPS	4 KBPS
0.05	0.026	5225	1482	0.051	2612	741	0.10	1340	380
0.10	0.053	2543	721	0.11	1272	361	0.21	638	181
0.15	0.081	1649	469	0.16	825	234	0.33	406	115

Notes: (1) Capacity calculated assuming 15 bit abbreviated result, 20 ms packet time, in 50% activity factor.

(2) Capacity (or burst rate) in Mbps.

If x is the number of retransmission attempts made on a given packet, the channel load G will be related to the offered load U by:

$$G = U(1 + E(x)).$$

Assuming the success of the first attempt and subsequent retrys are independent with probability q, the random variable x will have distribution

$$P(x) = \begin{cases} (1 - q)q^x, & 0 < x < n \\ q^n \quad , & x = n. \end{cases}$$

It follows that:

$$E(x) = \frac{q}{1-q} (1 - q^n),$$

and hence

$$G = U(1 - q^{n+1})/(1 - q)$$

or

$$G = U(1 - P_L)/(1 - P_L{}^{\frac{1}{n+1}})$$

The relationship between single-attempt loss q and channel traffic was previously established for the three ALOHA-type protocols. Hence, we have the following results (shown only for the $N\to\infty$ limit):

PURE ALOHA

$$P_L = (1 - e^{-2G})^{n+1},$$

or, eliminating G, we have

$$U = -\frac{1}{2}\,\ell n\left(1 - P_L^{\frac{1}{n+1}}\right)\left(1 - P_L^{\frac{1}{n+1}}\right)(1 - P_L)^{-1}.$$

as the relationship between performance P_L and offered load or utilization. By eliminating P_L we obtain the relationship between channel traffic attempts and utilization:

$$U = G\,e^{-2G}\left[1 - (1 - e^{-2G})^{n+1}\right]^{-1}.$$

SLOTTED ALOHA

Similarly, for slotted ALOHA:

$$P_L = (1 - e^{-G})^{n+1}$$

$$U = -\ln\left(1 - P_L^{\frac{1}{n+1}}\right)\left(1 - P_L^{\frac{1}{n+1}}\right)(1 - P_L)^{-1}$$

$$U = G\,e^{-G}\left[1 - (1 - e^{-G})^{n+1}\right]^{-1}.$$

The above relationships describe the full range of operation between no-retransmission ($n = 0$) and infinite-retransmission ($n = \infty$) or lossless systems. Figure 4.4.6 is a plot of utilization versus channel traffic for the slotted ALOHA cases and clearly shows the transition between stable operation for no-retransmission and unstable operation for the usual ($n = \infty$) slotted ALOHA, at around $n = 7$. For the likely area of voice operation at $n = 1$ or 2, slotted ALOHA is clearly stable.

Figure 4.4.7 is the companion plot of packet loss rate P_L versus utilization for the slotted ALOHA cases. The improvement in utilization at low P_L by allowing a few retrials as well as the instability for too many retrials is clear.

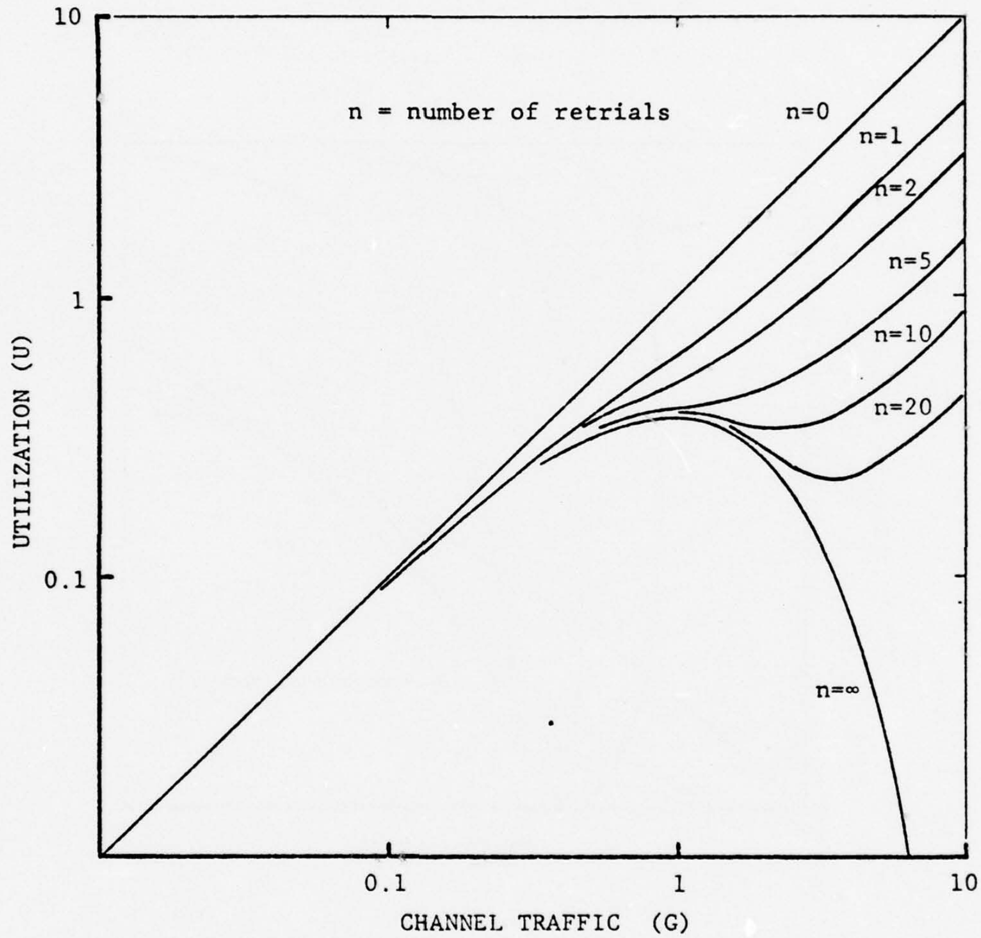


FIGURE 4.4.6 OFFERED LOAD AS A FUNCTION OF CHANNEL TRAFFIC OVER RETRANSMISSION POLICY RANGE FOR SLOTTED ALOHA

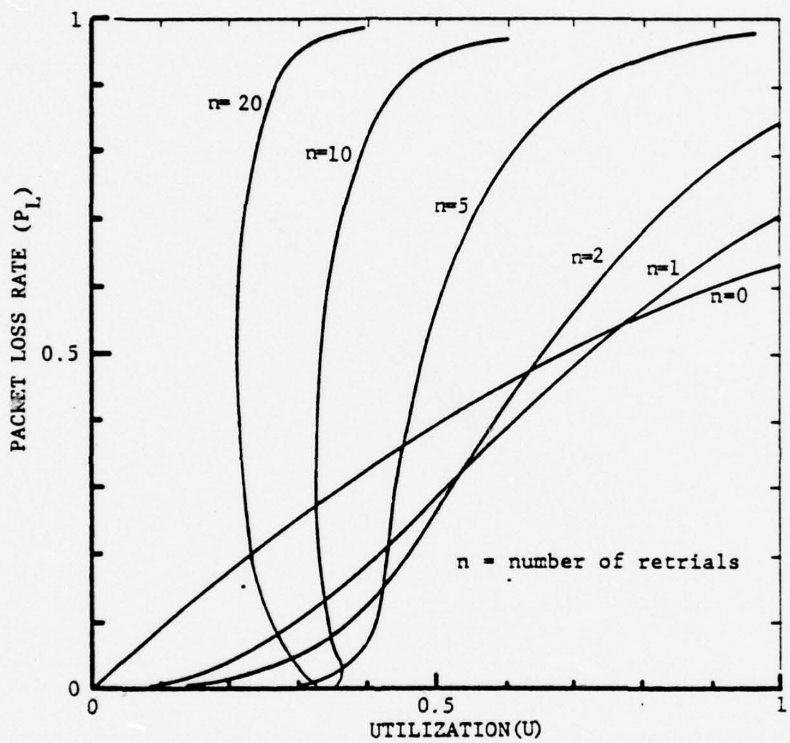


FIGURE 4.4.7 PACKET LOSS RATE AS A FUNCTION OF OFFERED LOAD OVER RETRANSMISSION POLICY RANGE FOR SLOTTED ALOHA

CAPTURE ALOHA

$$P_L = \left(1 - \frac{1-e^{-G}}{G}\right)^{n+1}$$

For CAPTURE, G cannot be explicitly eliminated:

$$U = G\left(1 - P_L^{\frac{1}{n+1}}\right)(1 - P_L)^{-1}$$

$$U = (1 - e^{-G})\left[1 - \left(1 - \frac{1-e^{-G}}{G}\right)^{n+1}\right]^{-1}$$

CAPTURE is stable for any n including n = ∞. The important effect for a packetized voice system is the very high efficiency for only a few retransmissions attainable. Figure 4.4.8 is a plot of packet loss rate vs. offered load for a CAPTURE system with n = 0, 1, 2. Only two retrys puts the offered load at 844 for 10% P_L. Making n small for CAPTURE guarantees a maximum delay and improves efficiency over making n = 0.

Comments

Table 4.4.2 compares the required burst rate for slotted ALOHA and capture ALOHA at a 10% packet loss rate, for 16 kbps packetized voice. For n = 2 slotted ALOHA requires slightly higher burst rate than the FVDA requirement, capture ALOHA requires a smaller burst rate for n > 0.

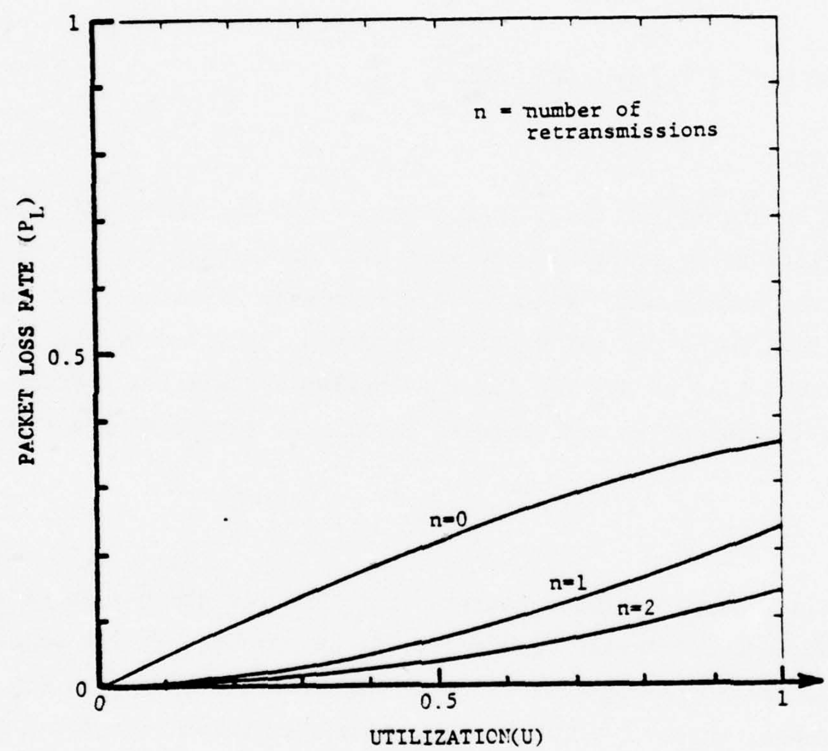


FIGURE 4.4.8 PACKET LOSS RATE AS A FUNCTION OF OFFERED LOAD OVER RETRANSMISSION POLICY RANGE FOR CAPTURE ALOHA

TABLE 4.4.2 EFFICIENCY AND BURST RATE REQUIREMENTS FOR FINITE RETRANSMISSION FOR SLOTTED ALOHA AND CAPTURE ALOHA

	n=0	n=1	n=2
Slotted ALOHA			
Utilization	0.105	0.290	0.370
Burst (Mbps)	1272	462	362
Capture ALOHA			
Utilization	0.210	0.616	0.844
Burst (Mbps)	638	217	159

Conditions: 16 kbps, 50% duty voice, 8000 Erlangs.

Note: For comparison, FVDA requires 243 Mbps.

4.4.6.3 Multiple Copy ALOHA

As an alternative protocol to finite retransmission protocols we now consider an ALOHA-type protocol where no retransmissions are allowed but multiple copies of each packet are sent. Clearly the collisions will increase over single-copy no-retransmission, but only one copy of a multiple-copy packet burst needs to survive.

If q is the loss or collision probability for a single packet copy, then sending n extra copies (at randomized times) will give a total packet loss rate of

$$P_L = q^{n+1},$$

since all (n+1) copies must be lost to produce a real or net packet loss. The channel traffic G is increased by the added copies over the offered traffic U:

$$G = (n+1)\ U.$$

To evaluate P_L for pure, slotted and capture ALOHA systems, the previously developed relationships relating q to channel traffic are substituted. In general g = 1 - S/G.

<u>PURE ALOHA</u>

$$q = 1 - e^{-2G}$$
$$= 1 - e^{-2(n+1)U}$$

Thus,

$$P_L = [1-e^{-2(n+1)U}]^{n+1}$$

or, equivalently,

$$U = \frac{-1}{2(n+1)} \ln(1-P_L^{\frac{1}{n+1}}).$$

SLOTTED ALOHA

Similarly, for slotted ALOHA,

$$P_L = [1 - e^{-(n+1)U}]^{n+1}$$

and

$$U = \frac{-1}{(n+1)} \ln(1-P_L^{\frac{1}{n+1}}).$$

CAPTURE

For a CAPTURE satellite, the single copy loss role is given by

$$q = 1 - \frac{1-e^{-G}}{G}$$

$$= 1 - \frac{1-e^{-(n+1)U}}{(n+1)U}.$$

Hence,

$$P_L = [1 - \frac{1-e^{-(n+1)U}}{(n+1)U}]^{n+1}.$$

The loss rates for multiple-copy slotted ALOHA and CAPTURE are plotted in Figures 4.4.9 and 4.4.10, respectively. As with the corresponding n-retransmission policies (Figures 4.4.7 and 4.4.8) we see that for a given small P_L, some improvement in utilization is possible with a multiple-copy, no-retransmission policy over single-copy, no-retransmission policy. The improvement is not as dramatic, however, as for n-retransmissions. Figure 4.4.11 is a comparison of utilization NS the policy parameter n for slotted ALOHA for both n-copy no-retransmission

and single-copy n-retransmission. The multiple-copy policy is better only than n=o (no retransmission). Thus (at a P_L of .1) if the extra delay of even one retransmission is not acceptable for voice the efficiency over single-copy can be increased from about 10% to 211% by sending 5 copies of every packet. Similarly, for a CAPTURE multiple copy system, the utilization can be increased to about 0.5 for a P_L of 0.1, by sending about 5 copies and using no retransmission.

4.4.6.4 Spread-Spectrum RMA

The multiple-access capability of spread-spectrum systems, alternatively referred to as CDMA, was discussed in some detail in Section 5.2.2.3 of the Task 1 report to DCA, December 1976. Based on the calculations made in that section (Figure 5.2.7), a bandwidth-constrained short code (128 chips) can be used to support approximately 10-20 multiples accesses, depending on the E_n/N_o required. It appears possible to apply spread spectrum of this type to RMA such as ALOHA and SLOHA systems.

The ALOHA systems, because of their bursty mode of operation, require a ground ET with high peak-power capability. The application of spread spectrum would reduce this peak-power requirement (the average power would be identical, however) while providing some multiple-access capability. System analysis regarding ALOHA with spread spectrum has been performed by Chiao [14b]. The resulting throughput channel traffic relationship for the system with retransmission is:

$$S = G \sum_{i=0}^{L} (2G)^i \frac{e^{-2G}}{i!}$$

where L is the additional number of multiple-access channels via spread spectrum. Equation (1) is plotted in Figure 3.3.12. Note that, even with only one additional channel (L = 1), the throughput, S, is increased from 18% of ALOHA to 40%.

For the system without retransmission, which is more suitable for voice communications, the probability of packet loss is:

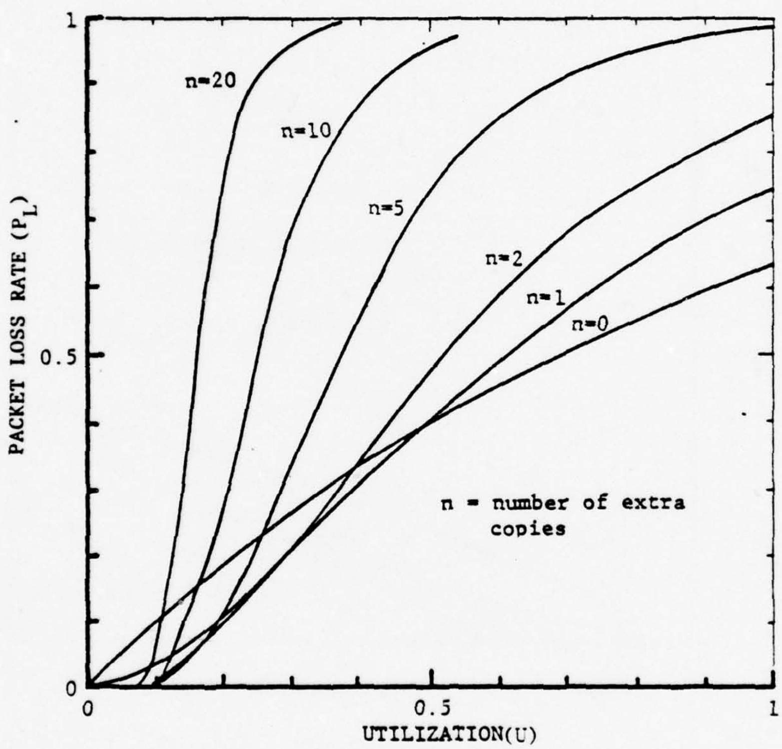


FIGURE 4.4.9 PACKET LOSS RATE FOR MULTIPLE COPY NO-RETRANSMISSION ALOHA

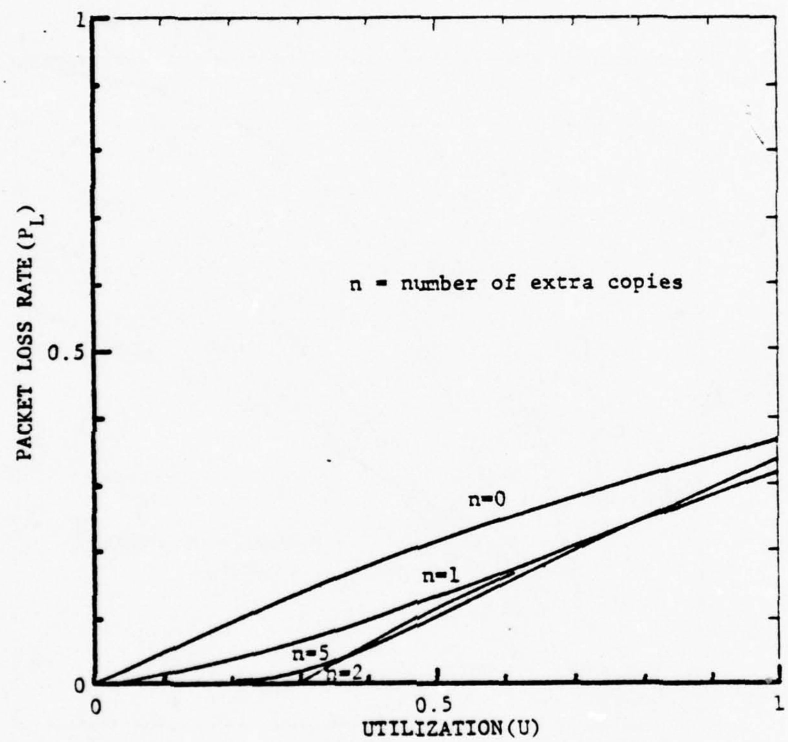


FIGURE 4.4.10 PACKET LOSS RATE FOR MULTIPLE COPY CAPTURE ALOHA

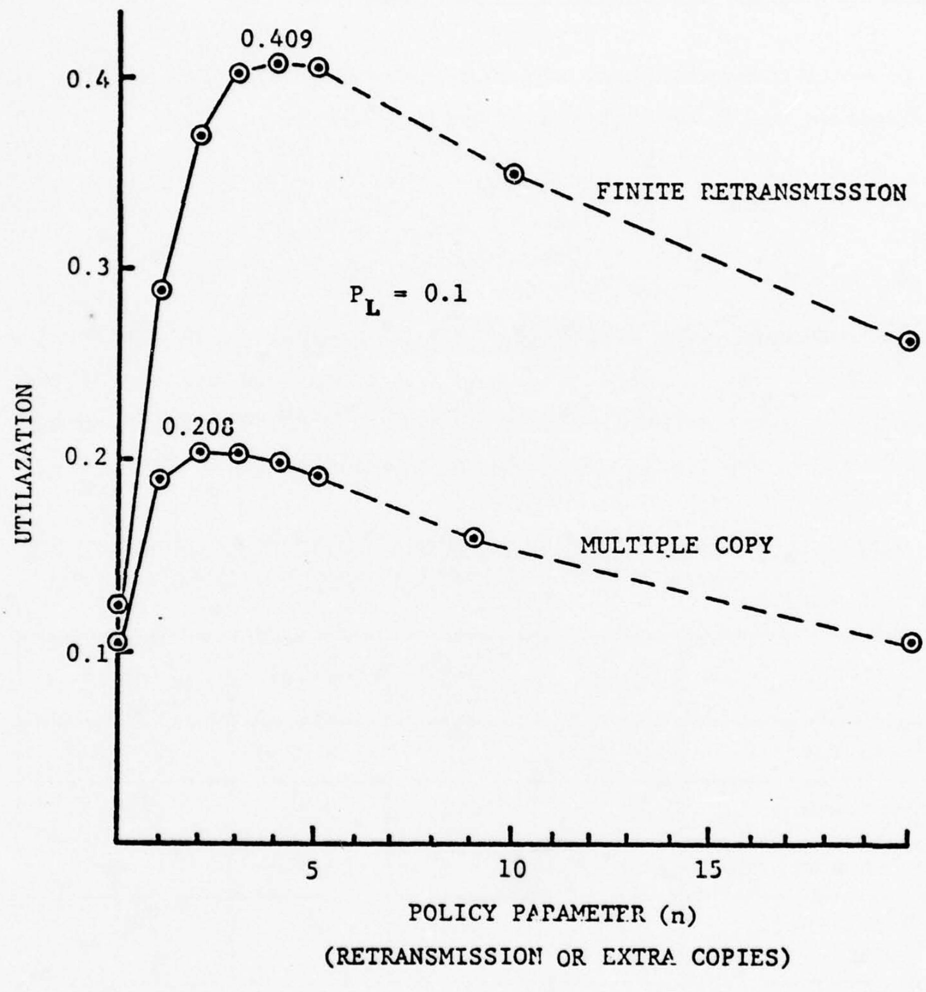


FIGURE 4.4.11 EFFICIENCY OF FINITE RETRANSMISSION AND MULTIPLE-COPY ALOHA POLICIES

$$P_L = 1 - \sum_{i=0}^{L} (2U)^i \frac{e^{-2U}}{i!}$$

Spread-Spectrum ALOHA without Retransmission

If voice communications are to be served by SS-ALOHA without retransmission, the probability of a lost packet is

$$P_L = 1 - \sum_{i=0}^{L} (U)^i \frac{e^{-U}}{i!}$$

The analyses given above, together with other known results (see Figure 4.4.13), can be used to assess the system performance of the following four RMA schemes serving a traffic model requiring an aggregate of 128 Mbps of data rate. The results are tabulated in Table 4.4.3.

TABLE 4.4.3 RMA SYSTEMS ASSESSMENTS (WITHOUT RETRANSMISSION) DATA RATE - 128 MBPS AT PACKET LOSS RATE $P_L = 0.1$

	ALOHA	REGULAR SLOHA	SLOHA-SS	SLOHA-MULTIPLE COPIES
Base Rate	128 Mbps	128 Mbps	9.14 Mbps	5.2 Mbps
Efficiency	0.053	0.11	0.11	0.21
Data Burst Rate	2430 Mbps	1215 Mbps	1170 Mbps	620 Mbps
Packet Efficiency	69%	48%	48%	48%
Burst Rate with Overhead	3539 Mbps	2529 Mbps	2432 Mbps	1291 Mbps
Required (G/T + EIRP) at K_u Band with 13 dB (E_b/N_o + Margin)	85.3 dBW	83.5 dBW	83.3 dBW	80.55 dBW
Relative Complexity of Terminal	Least	Moderate	Most	More

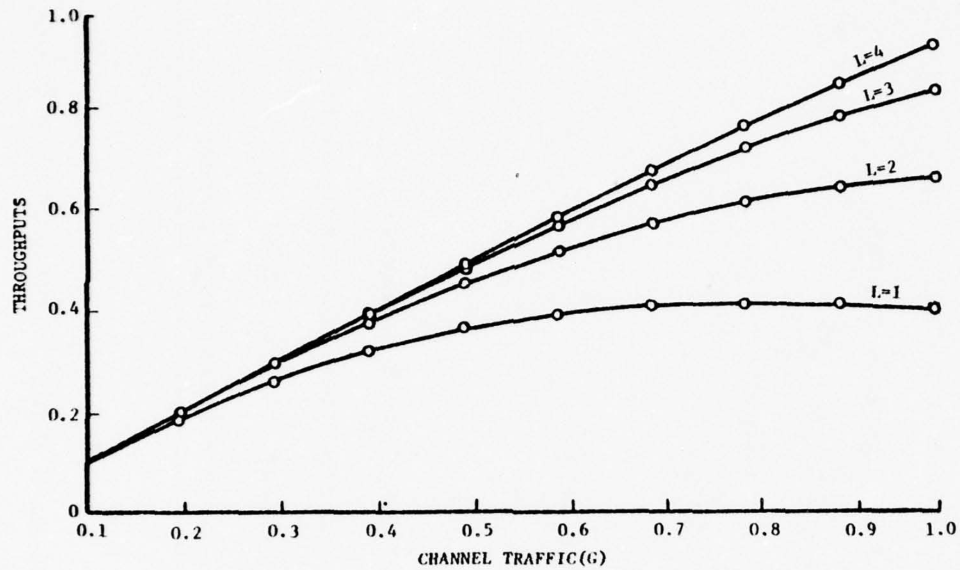


FIGURE 4.4.12 ALOHA WITH SPREAD SPECTRUM

given $\left\{\begin{array}{l} P_L \\ L \end{array}\right.$ (probability of lost packet)

(number of simultaneous channels

⇩ via $P_L = 1 - \sum_{i=0}^{L} U^i \frac{e^{-U}}{i!}$

S (channel utilization)

⇩ via $R_b = \lambda\tau/S$

R_b (data rate)

given g (processing gain)

⇩ via $R_c = g\, R_b$

R_c (chip rate)

⇩ via $C/kT = E_b/N_o + R_c +$ margin

C/kT (required carrier-to-noise-density ratio)

⇩ via $G/T = C/kT + L + k - EIRP$

G/T (required earth terminal sensitivity)

FIGURE 4.4.13 SS-ALOHA SYSTEM CALCULATION WITHOUT RETRANSMISSION

4.4.7 Mixed Voice/Data Packet ALOHA Protocols

The distinctly different requirements of voice and data packets can be summarized as follows:

VOICE (duplex call)

- Call holding time 2-5 minutes
- About 10% loss rate nominally acceptable
- Delay of more than 1 or 2 round trips (.28 sec) not normally acceptable

Data Packets

- Isolated packets
- No significant loss rate acceptable
- Larger delays tolerable, nominally ≃ 10 round trips

These different characteristics suggest that for a mixed voice/data common-user packetized ALOHA-like RMA system different kinds of packets should be labeled to receive different kinds of treatment as follows:

Voice Call Protocol:

- Call setup between earth terminals to allow abbreviated voice packet header - identifying only particular call-in-progress
- Finite retransmission protocol (0-2 retransmissions)

Isolated Data Packet Protocol:

- Full routing information headers
- Non-reservation, infinite-retransmission ALOHA

Packet String Message Protocol:

- Reservation protocol (e.g., Robert's Reservation on Round-Robin Reservation)

Since both the finite retransmission and reservation parts of this mixed protocal are stable, even for a non-CAPTURE satellite, it seems plausible that a small fraction of infinite-retransmission isolated-data packet traffic could be added without destroying the stability. At any rate a CAPTURE satellite would insure stability.

We remark that no precise performance results on this mixed protocol are available because direct analytical attack is not apparently tractable. A simulation approach can easily be envisioned to study performance, but such an approach is beyond the scope of the present study.

Finally we remark that while this mixed protocol provides for the needs of both voice and data <u>packets</u> it does not permit carrying of any true stream data where bit stream synchronization is maintained, e.g., for FAX, digital television, or conventional (stream-oriented) COMSEC equipment.

4.4.8 Directional Variable Random Access

We now propose and analyze a new RMA technique called Directionally Variable Random Access (DVRA) which extends the directionally variable demand access (DVDA) approach (circuit-switched) to a packet-switched concept. No central access control is required.

Let the total capacity of the satellite C be split into N subchannels (one subchannel per earth terminal). Each ET contains a buffer where arriving packets are assembled and queued for transmission over the allocated fraction of the broadcast channel at rate C/N. To receive packets, every ET reads the header bits of all packets being broadcast by all other ETs. There is no contention, collisions, or control (other than timing if TDMA is used to divide the total capacity into N subchannels.)

The performance of such a system degrades with load in two ways: delay of packets in the buffer, and lost packets when the buffer overflows. An arbitrarily large buffer allows efficiency to be arbitrarily near 100% without packet loss. However, the resulting large random delays of packets in the transmit buffer is not acceptable for voice. If the buffer size is made small (B bits) the maximum delay is limited to $B \cdot N/C$.

C must be greater than total offered load (128 Mbps for 16 kbps voice ditization). Thus for a 16 kilobit buffer, the maximum delay is less than $.128 \times 10^{-3} \times N$ seconds. Thus for up to 200 earth terminals buffer delay is less than one propagation trip.

<u>Packet Loss Rate</u>

The transmit buffer will hold

$$n = B/L$$

packets awaiting transmission, where L is the packet length. The buffer packet queue length equilibrium distribution is given (using independent arrivals and departures) as

$$P_j = \begin{cases} Q^j P_o , & o \le j \le n \\ Q^n (1-U/N) P_o , & j=n \end{cases}$$

where

$$Q = \frac{U(N-1)}{N-U}$$

and

$$P_o = \frac{1-Q}{1-Q^n[1-(1-Q)(N-U)/N]}$$

Arriving packets find a full buffer with probability P_n and are hence lost at that rate. For large N, the above reduces to a Packet loss function of

$$P_L \simeq \frac{U^n(1-U)}{1-U^{n+1}}$$

This loss function is plotted in Figure 4.4.14. Note that the maximum loss rate as $U \to 1$ is given by

$$\max P_L = \frac{1}{n+1}$$

For the 32-packet buffer (lowest curve) the loss rate is limited to about 3% and is down to less than 2 packets in 10,000 at a utilization of .8. The 32-packet buffer corresponds to a 16384 bit buffer for packets of size 512 bits.

At the traffic levels of this study the DVRA technique requires only about half the capacity of the best circuit switching (FVDA). This doubling in efficiency is due to its ability to fully utilize the assumed 50% voice duty factor.

For a buffer of the size above or larger, the loss of packets is a sudden or threshold effect. By operating at a lower-than-threshold level, non-voice data packets and messages can be handled along with voice in an essentially lossless mode. Alternately, the buffer space could be segregated between data and voice, with voice receiving perhaps 9 packets of space (max P_L = .1) and data receiving perhaps 50 packets of space. By operating at no more than about 90% utilization and giving the small % data load higher priority than most voice, the data operation would be essentially lossless. The very few losses which would occur for data packets can be deleted and retried by end-to-end protocol.

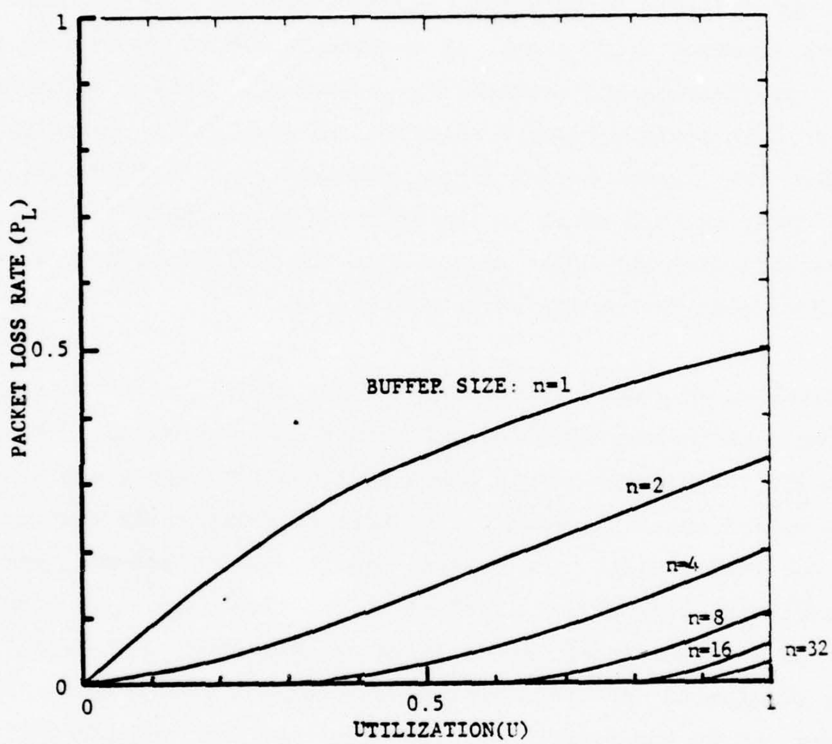


FIGURE 4.4.14 PACKET LOSS RATE FOR DIRECTIONALLY VARIABLE RANDOM ACCESS PROTOCOL

We also remark that like circuit-switched DVDA, the high efficiency of DVRA depends on the assumed traffic split among earth terminals matching the corresponding allocated fractional satellite capacity. There exist techniques which allow the fractional split to be changed fast enough to track traffic changes in real time. For example, the MAT-1 approach [52] is to shift the partitions in a TDMA frame. The fraction of frame used by each ET up-link should be matched to total traffic originating and terminating at that ET. Such a scheme apparently requires both traffic level monitoring and (centrally) coordinated control. The control action is not required on a call-by-call or packet basis - only at such times as the split is to be changed. It has been demonstrated with the MAT-1 system that the TDMA frame boundaries can be shifted without disturbing calls-in-progress.

Finally, we remark that rather than a single buffer where all arriving packets are assembled and queued FIFO, an actual DVRA terminal in the DCS environment should use a multiple priority queue system with enough buffer space reserved for highest priority calls that no losses occur for those calls. The present design outline and analysis is thus not realistic or complete. Because of the flexibility and higest efficiency of all techniques investigated in this study we believe further study, simulation and trade-offs of this access method beyond this study is important to the future DCS. The next section outlines how a hybrid circuit/packet system can evolve toward primarily utilizing this new DVRA technique.

4.5 SUMMARY OF DAMA/RMA EFFICIENCY/PERFORMANCE TRADEOFFS

4.5.1 Voice/Traffic Requirements Summary

The emphasis and conclusions in this study are dominated by the requirements of the projected common-user DCS traffic model of the 1985-1995 time frame, and the communication requirements of digitized and/or packetized voice.

The salient traffic model parameters are:

~400,000 voice terminals

~8,000 Erlangs originating load (beyond PBX)

The 8,000 Erlangs translates to 256 Mbps of total data rate at 16 Kbps voice digitization, or 64 Mbps @ 4 Kbps. Compared to an aggregate 9.6 Mbps of aggregate nonvoice requirements, it is clear that:

- Voice Predominates

Since the voice digitization rate directly scales the total data rate requirements for circuit-switched DAMA techniques and nearly so for packet-switched DAMA techniques we have chosen two baseline alternatives of 16 Kbps (CVSD) as a representative of wideband techniques and 4 Kbps as a representative of the narrowband PEV class of Vocoders.

Since two-way voice communication is intolerent of variable or long fixed delays of more than about 1/2 second, RMA techniques which lose packets rather than retry until successful are taken as appropriate. While not yet definitively established as acceptable, a packet loss rate for packetized voice of 10% is taken as a baseline nominal figure for comparison purposes.

Packetized voice was assumed to be generated only when a talker is active, thus reducing the total baseline voice requirements by the duty factor - assumed to be 50%. (This is conservative - see Chapter 2.)

4.5.2 Efficiency/Capacity Comparisons

The efficiency or utilization of the circuit-switched DAMA techniques is equivalent to occupancy:

$$\text{Efficiency} = \frac{\text{total load}}{\text{total number of circuits}} = U$$

at a specified blocking (5% is used). If C_o denotes the base capacity required (depending on the digitization rate) then the total satellite bit rate capacity required is:

$$C = C_o/U$$

Similarly, the efficiency or utilization U for packet-switched RMA techniques is defined by the above relationship.

Table 4.5.1 gives the base rates (C_o) for the two digitization rates and two basic types of switching in this study.

Table 4.5.2 compares utilization and the corresponding total capacity required C for the alternative DAMA/RMA techniques for both 4 and 16 Kbps digitization of the (8,000 Erlang voice) load. The required capacity shown does not consider additional implementation overhead.

The clear conclusions of this comparison are:

- The most efficient packet-switched technique is DVRA with a total required capacity of 32 (128) Mbps for 4 (16) Kbps packetized voice.

TABLE 4.5.1 BASE RATES (MBPS)

TYPE	VOICE DIGITIZATION RATES	
	4 KBPS	16 KBPS
Packet	32	128
Circuit	64	256

TABLE 4.5.2 UTILIZATION/CAPACITY COMPARISONS FOR VOICE RMA AND ALTERNATIVE DAMA TECHNIQUES

	U	C-4KB	C-16KB
Packet RMA			
0-retry P. ALOHA	0.053	604	2415
1-retry P. ALOHA	0.14	229	914
0-retry S. ALOHA	0.11	291	1164
1-retry S. ALOHA	0.29	110	441
3-copy S. ALOHA	0.21	152	610
0-retry S.S. ALOHA	0.11	291	1164
0-retry C. ALOHA	0.21	152	610
1-retry C. ALOHA	0.62	52	206
6-copy C. ALOHA	0.50	64	256
Destination Var. RA (DVRA)	≃1.0	32	128
Circuit-DAMA			
Terrestrial (Reference) *	0.34	188	752
FAMA (Reference) (N=70)	0.46	139	557
DVDA (N=70)	0.94	68	272
FVDA (N=70)	1.05	61	244

* From Polygrid Network Model in Chapter 3 (23,680 total two-way circuits).

- The most efficient circuit-switched techniques are FVDA requiring 61 (244) Mbps and DVDA requiring 68 (272) Mpbs satellite capacity for bit-stream 4 (16) Kbps digitized voice.
- The ALOHA-based RMA techniques are not efficient enough to be used except for CAPTURE ALOHA which requires a new type of processing satellite. Even with the required more expensive and risky space segment CAPTURE ALOHA is not as efficient as DVRA.

5 COMMUNICATION SATELLITE REALIZATION ISSUES

5.1 LINK PERFORMANCE

The performance of the ET-satellite/ET-communication link as measured by the trade between bit error rate and data rate (burst rate) is normally limited by the down link parameters. The limiting parameters of the downlink are either signal-to-noise ratio at the receiver demodulator (power limited) or the transponder bandwidth (bandwidth limited). The maximum burst rate that can be passed through the satellite link will be defined in this report as the capacity, denoted by the symbol C:

C = minimum (bandlimited burst rate, power limited burst rate)

Capacity as a function of satellite EIRP + earth terminal G/T is illustrated in Figure 5.1.1. The limiting factors on satellite capacity are discussed in the following subsections.

5.1.1 Capacity Limits

i. Bandwidth Limited Case:

The link equation expressed in dB is:

$$R_b = W + B - C_W \qquad (5.1.1)$$

where

R_b = burst rate

W = satellite transponder bandwidth

B = bit-rate-to-symbol-rate ratio (0 dB for BPSK, 3 dB for QPSK, etc.)

C_W = ratio of W to bandlimited symbol rate through the transponder (typically 1.2 (0.8 dB)).

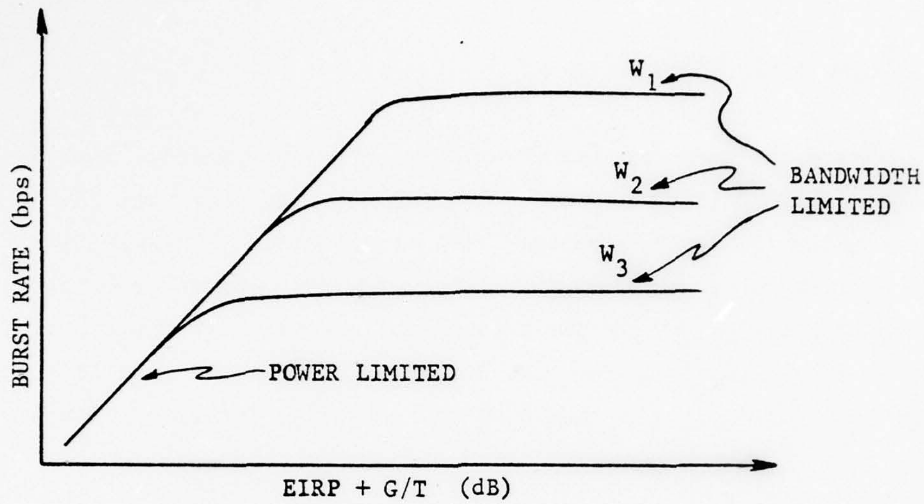


FIGURE 5.1.1 CHANNEL CAPACITY OF COMMUNICATIONS SATELLITES

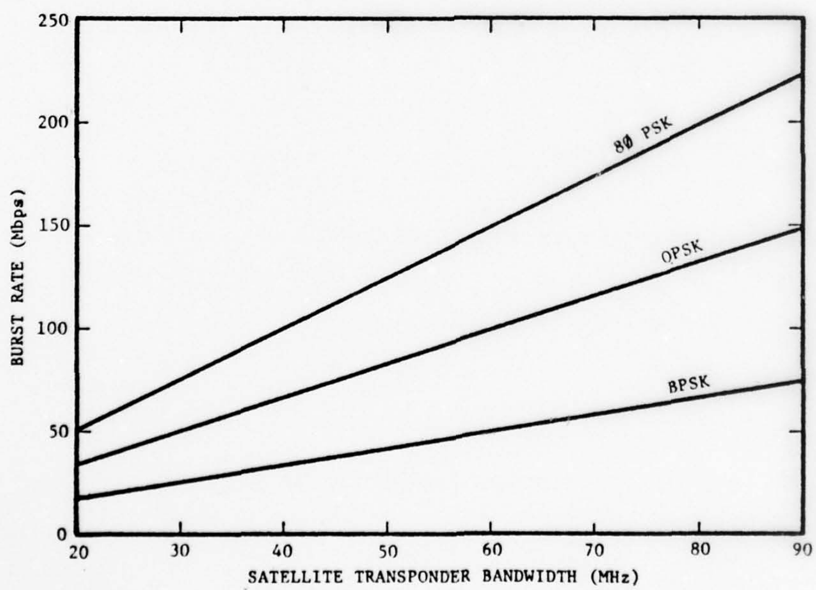


FIGURE 5.1.2 CHANNEL CAPACITY OF BAND LIMITED CASE

The link equation above is plotted for R_b vs W in Figure 5.1.2.

ii. Power Limited Case:

The link equation expressed in dB is:

$$R_b = EIRP + G/T - L_S - k - (E_b/N_o) - L_M \qquad (5.1.2)$$

where

EIRP = satellite transponder radiated power including transmitting antenna gain

G/T = receiving terminal antenna gain/noise temperature ratio

L_S = space loss at carrier frequency and distance to satellite

k = Boltzman's constant (-228 dBW/Hz°K)

E_n/N_o = bit-energy-to-noise-density ratio for a given bit error rate (BER)

M = system margin to account for miscellaneous losses

Link equation is plotted in Figure 5.1.3 for C, X, and K_u bands as shown.

5.1.2 BER vs. Modulation and E_b/N_o

The bit error rate (BER) is a measure of performance of digital communications. BER is a function of modulation techniques, receiving conditions and noise statistics. Digital modulations include techniques such as on-off keying, FSK, PSK, etc. In space communications, coherent PSK and differentially coherent PSK are most frequently considered. A relatively new constant envelope modulation technique known as MSK (minimum shift keyed) is now also emerging in the digital space communication field. Figure 5.1.4 shows the relationships of BER vs. E_b/N_o for several modulations.

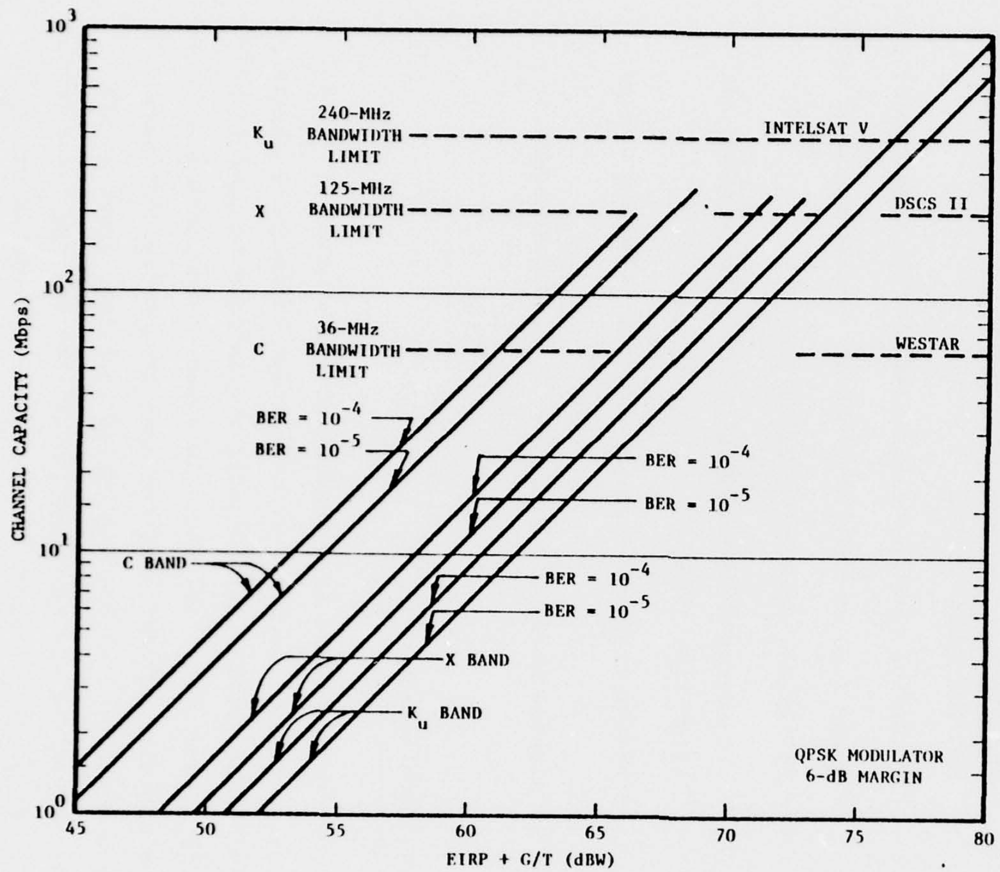


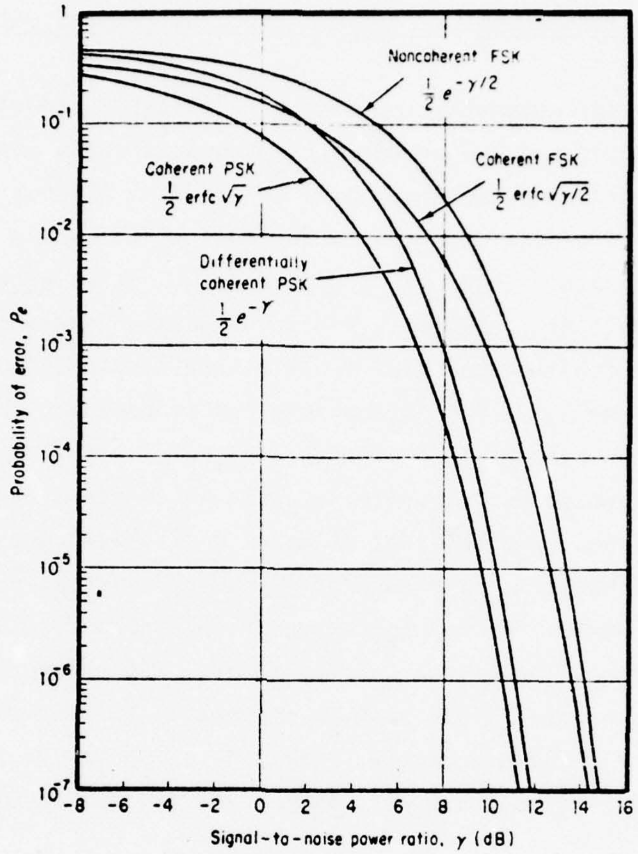
FIGURE 5.1.3 CHANNEL CAPACITY OF POWER LIMITED CASE

5.1.3 Link Relationships vs. Forward Error Correction Coding

For satellite channels, the most effective FEC (forward error correction) coding can reduce the E_b/N_o required for a given desired bit error rate by 5-6 dB or more compared to a system without control. The cost in terms of system constraints for adding FEC coding is, of course, bandwidth expansion. Thus, FEC techniques can not be effectively applied for a bandwidth limited system. For power limited systems, FEC can be used to reduce the required EIRP + G/T of the downlinks to derive substantial cost savings. FEC coding requires redundancy; but this redundancy need not reduce throughput of a power limited system if enough additional bandwidth is available. Actually, a small bandwidth increase can produce considerable power savings. For example, a rate-1/2 cost (100% redundancy) will require double the bandwidth of an uncoded system; while a rate-3/4 code (33% redundancy) having coding gains only about 1 dB less needs the bandwidth expansion only by a factor of 4/3. FEC coding can also be applied to terrestrial links such as microwave, HF and tropospheric in addition to space communications. Table 5.1.1 shows the relationships of bandwidth expansion, threshold E_b/N_o, and several modulation-coding methods.

5.1.4 Survey and Forecast of Satellite EIRP and Bandwidth

The satellite EIRP and bandwidth are two of the major resources of satellite communication systems. The sizes of EIRP and bandwidth are generally limited by the state-of-the-art of hardware technology and deployment mechanics.



From S. Stein and J. Jay Jones [67]

FIGURE 5.1.4 ERROR RATES FOR SEVERAL BINARY SYSTEMS

TABLE 5.1.1 MODULATION/FEC CODING TECHNIQUES

MODULATION	CODING	BW/DATA RATE RATIO	THRESHOLD E_b/N_o @ BER = 10^{-5}
PSK	No	2	9.6
PSK	Rate 3/4	2.67	7.3
PSK	Rate 1/2 (Viterbi decoder)	4	4.5
DPSK	No	2	10.3
DPSK	Rate 1/2 (Viterbi decoder)	4	6.5
QARK	No	2	10.3
MFSK (M = 16)	No	8	8.8

The transponder EIRP power has two major limiting factors:

EIRP limited by:
- transponder/antenna technology
- prime power supply
  - solar power
    - weight and solar array deployment mechanism limited
  - nuclear power
    - technology limited

The bandwidth is also limited by two major factors:

BW limited by:
- transponder/antenna technology
- frequency allocation
  - lower frequencies (C, X band)
    - smaller bandwidth
  - higher frequencies (K_u, K band)
    - higher space loss + rain loss

A recent COMSAT study suggests that, because of the tremendous growth of information transfer demand via satellite of recent years, satellite communications systems are being transitioned from essentially technology constrained power-limited systems to larger, environmental constrained BW-limited systems. This trend is further borne out by a DCA study on the common user traffic of the DCS community. This trend is pictorially illustrated in Figure 5.1.5. Note that this figure is speculative in nature and that it is not intended to be substantiated in any detail. To be more specific, Figures 5.1.6 and 5.1.7 are prepared to show the EIRP and frequency plans of major military systems, respectively.

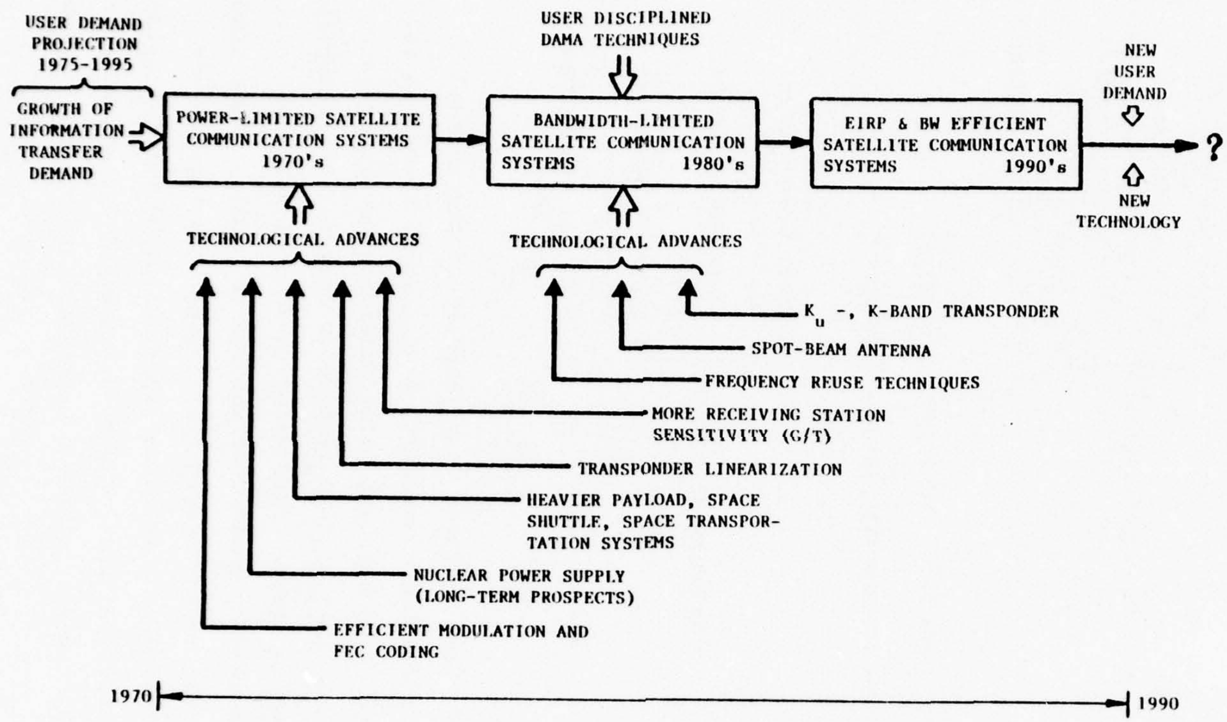


FIGURE 5.1.5 EVOLUTION OF EFFICIENT SATELLITE COMMUNICATION SYSTEMS

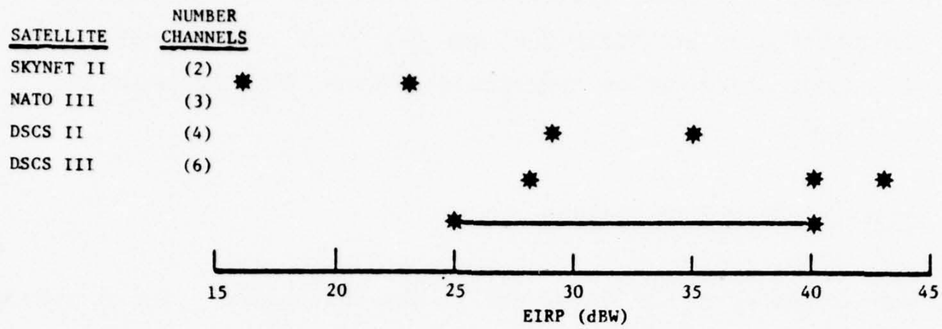


FIGURE 5.1.6 MILITARY COMMUNICATION SATELLITE COMPARISON

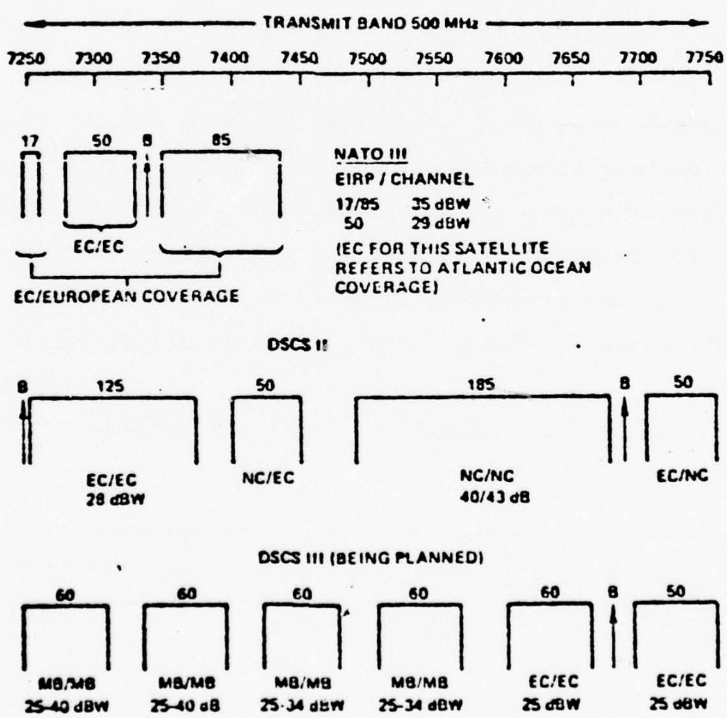


FIGURE 5.1.7 TRANSMIT FREQUENCY BANDS OF MILITARY COMMUNICATION SATELLITES

For commercial systems, typical EIRPs are shown in Figure 5.1.8 and the frequency plan of INTELSAT V, the most complicated commercial communication satellites to be operational before 1980, is shown in Figure 5.1.9 [61].

5.1.5 Reference Satellite Systems

In subsequent system analyses, it is important that a set of reference satellite systems be selected to reflect the current technical maturity and the impact of near-future systems. Three communication satellite systems are selected for this purpose. They are:

WESTAR	(mature C-band technology)
DSCS II	(mature X-band technology)
INTELSAT V	(developed K-band technology)

Parameters for the three are listed in Table 5.1.2. These three systems will be used as references against which capacity requirements of various DAMA techniques are compared. The capacity of each system can be evaluated using the bandwidth and power limit expressions in Section 5.1.1 and the parameters given in Table 5.1.2 below. The resulting capacities vs. earth station G/T are plotted in Figure 5.1.10.

TABLE 5.1.2 REFERENCE SATELLITE SYSTEMS

SATELLITE SYSTEM	FREQUENCY BAND	SPACE LOSS	EIRP	BW/TRANSPONDER	ANTENNA COVERAGE	MARGIN
WESTAR	C	196 dB	33 dBW	36 MHz	EC	6 dB
DSCS II	X	202 dB	28 dBW	125 MHz	EC	6 dB
INTELSAT V	K_u	204.5 dB	47 dBW	240 MHz	Spot beam (EC assumed in system analysis)	6 dB
E_b/N_o = 7 dB for BER = 10^{-3}, 9.6 dB for BER = 10^{-5}, QPSK modulation						

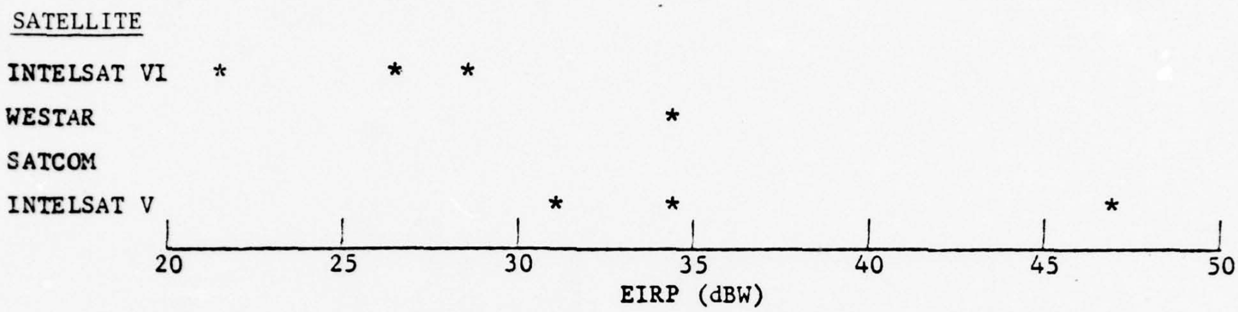


FIGURE 5.1.8 COMMERCIAL COMMUNICATION SATELLITE COMPARISON

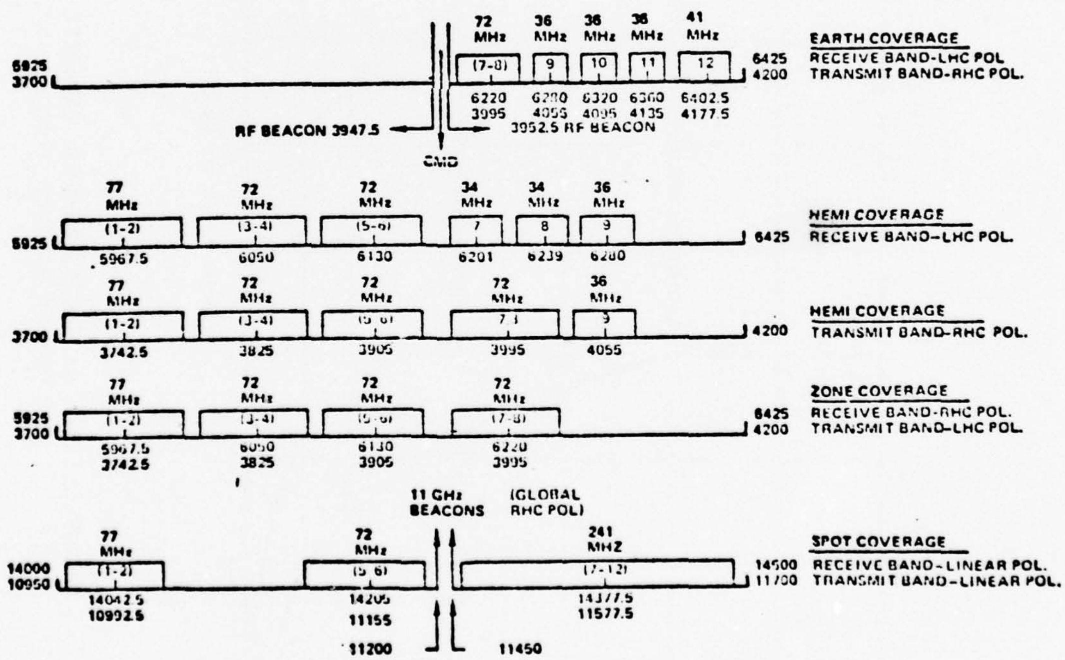


FIGURE 5.1.9 INTELSAT V FREQUENCY PLAN

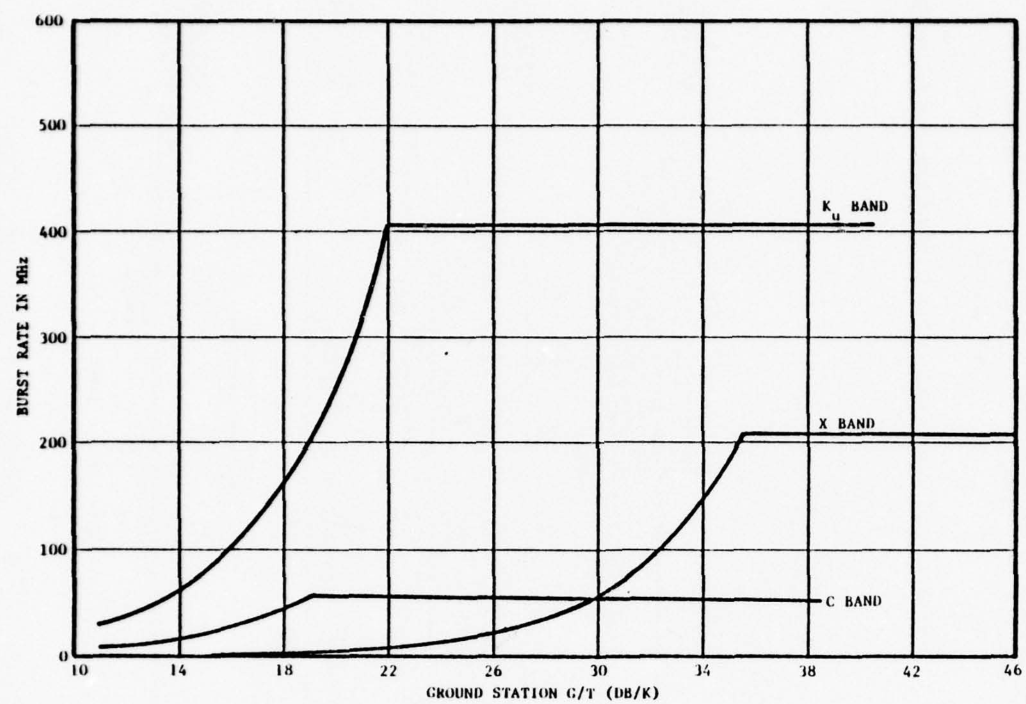


FIGURE 5.1.10 SATELLITE CHANNEL CAPACITY AS A FUNCTION OF GROUND STATION SIZE

5.2 CHANNELIZATION TECHNIQUES

5.2.1 Classification Channelization Techniques

5.2.1.1 OMA (Orthogonal Multiple Access) Techniques - Non-Contentional

In OMA, each channel is transparent to the others and there is no contention among channels. OMA techniques can be subdivided into three subclasses:

i) Time-division multiple access (TDMA)
ii) Frequency-division multiple access (FDMA)
iii) Code-division multiple access (CDMA)

5.2.1.2 RMA (Random Multiple Access) - Contentional

In this class, user pairs are given the opportunity to contend a link (from a pool of links, possibly) with others after they secure an access via a demand assignment discipline. In general, RMA involves two levels of contention. To assess the relative merits of such representative systems, a comparison of some key features of TDMA, FDMA, CDMA and RMA is made in Table 5.2.1. It appears that the throughput efficiency of DAMA systems is proportional to, among other things, the degree of complexity of network control required by the system. The selection of a DAMA system for a given user community thus represents an optimization problem requiring trade studies of many parameters, among them:

A. Space segment: EIRP, G/T, bandwidth, antenna beams
B. Ground segment: EIRP, G/T, antenna size
C. Netting efficiency and delay
D. Degree of network control and vulnerability
E. Impact of mixture of user stations.

TABLE 5.2.1 COMPARISON OF MULTIPLE-ACCESS TECHNIQUES

TYPE	TECHNIQUE	ADVANTAGES	DISADVANTAGES
FDMA	One frequency band assigned to each channel.	1. No network timing required. 2. Flexible control in matching downlink to terminal size (by varying power).	1. Requires uplink power control or multiple channel satellite transponder with separate limiters. 2. Susceptible to intermodulation caused by both active and passive circuit components. 3. With multiple carriers in single satellite transponder, requires complex frequency management in real time for maximum number of channels.
TDMA	One time slot as assigned to each channel.	1. Not susceptible to intermodulation. 2. Single channel satellite transponder operated at saturation. 3. Bandwidth efficient.	1. Requires network timing. 2. Less flexibility in matching downlink to terminal size (accomplish by varying data rate in discrete steps). 3. Buffer storage required.
CDMA	One code assigned to each channel. All channels on same frequency.	1. Jam resistant. 2. Easy transition from unstressed to stressed environment. 3. Flexible control in matching downlink to terminal size (by varying power).	1. Code acquisition required. 2. Bandwidth inefficient. 3. Self noise limits maximum number of channels 4. Requires uplink power control.
RMA	One or several wideband channels shared by all users on contention basis.	1. Minimal control or no control required. 2. Jam resistant. 3. Cost effective for low duty cycle common users.	1. Performance degrades rapidly as traffic intensity increases. 2. Low throughput. 3. Throughput can be increased at the expense of introducing more control.

5.2.2 Orthogonal Multiple-Access Techniques [19, 52, 62, 63, 64]

5.2.2.1 TDMA

A TDMA system consists of dividing a time frame into many smaller non-overlapping time slots. In any time slot, only one carrier accesses the satellite. In TDMA, usually one station acts as reference and sends periodic bursts without closed-loop control. The other stations in the network use closed-loop synchronization through the satellite to place its burst transmissions within their time slots (a guard time is provided between time slots to cushion timing uncertainties due to satellite motion, slant range differences, etc.). The burst lengths are not necessarily the same, since different traffic origination loads may be transmitted by different stations. Reconfiguration of burst lengths at each station to accommodate traffic variations can be accomplished manually or by using a microprocessor.

The number of accesses for TDMA is, typically, 5-15, but up to 60 accesses can be accommodated with current technology.

To assess the channel capacity of TDMA, the following formulae are available:

i) Bandwidth-limited case:

$$R_b = W + B - C_W \quad \text{(dB)} \tag{5.2.1}$$

where R_b = transmission bit rate
W = satellite transponder bandwidth
B = bit-rate-to-symbol-rate ratio
C_W = ratio of W to band-limited symbol rate through the transponder (typically, 1.2).

ii) Power-limited case:

$$R_b = EIRP + G/T - L_s - k - (E_b/N_o) - M \qquad (5.2.2)$$

where EIRP = satellite transponder EIRP

G/T = receiver G/T

L_s = space loss

k = Boltzman's constant (-228 dBW/Hz°K)

E_b/N_o = bit-energy-to-noise-density ratio for given BER

M = system margin.

If the power-limited channel capacity is less than that of the bandwidth-limited, forward error correction coding can be applied to provide 2-5 dB improvement.

To obtain an equivalent number of channels in terms of channel capacity, the following formula can be used:

$$N = \frac{1}{V}\left(R_b - \frac{m \cdot P}{T}\right) \qquad (5.2.3)$$

where
N is the number of channels
V is the bit rate for one channel
m is the number of multiple accesses
P is the number of preamble bits
T is the frame period.

Equation (5.2.3) is plotted in Figure 5.2.2 using the following values for convenience:

R_b = 60 MHz

P = 150 bits

T = 750 μsec

Both the 16-kbps channels and the 64-kbps channels are presented. By equating N = m in Eq. (5.2.3), the number of accesses for single channel per burst for both cases can be obtained. Figure 5.2.3 shows that the multiple-access capability of TDMA can be increased only at the tremendous expense of communication efficiency (number of channels). It appears that TDMA is more suitable for the medium-to-heavy traffic handling with the number of accesses no more than 50 and the number of channels per access no less than 20 or so.

Some of the advantages and disadvantages are already highlighted in Table 5.2.1. To reiterate, Table 5.2.2 is prepared to show some of the system impacts of TDMA. Looking into the future, TDMA presents technical challenges at least on two fronts:

A. Implementation of burst rate beyond 500 Mbps
B. Adaptation to satellite-switched, spot-beam TDMA.

5.2.2.2 FDMA

A FDMA system consists of dividing a frequency band into many smaller non-overlapping frequency bands. At any frequency band, only one carrier accesses the satellite. There are several transmission impairments that must be considered in designing FDMA systems to minimize intermodulation, intelligible crosstalk, and other interference effects. These impairments are limited to acceptable levels by backing off the operating point of the nonlinear TWT. However, this results in less channel capacity as compared to a single-access mode. Additionally, other general constraints such as uplink power control, frequency coordination, and vulnerability to interference must also be considered in FDMA system design. There are three basic types of FDMA being used or under development at this time. They are listed in Table 5.2.3.

Actual system capacity calculation for FDM-FM-FDMA is quite tedious because of the backoff, group delay, and the nonlinear nature of the FM

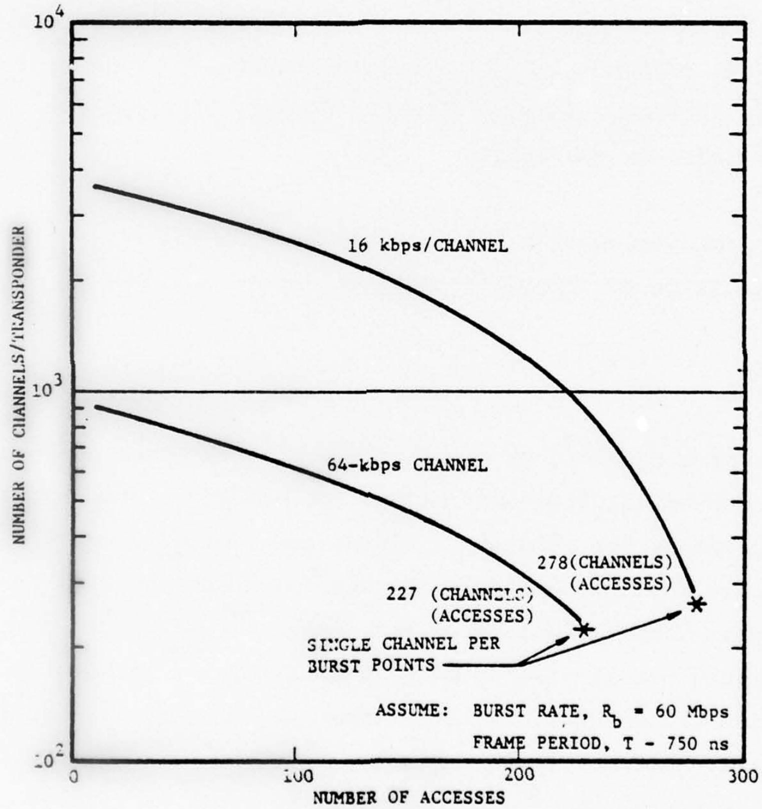


FIGURE 5.2.2 NUMBER OF CHANNELS AS A FUNCTION OF NUMBER OF ACCESSES

TABLE 5.2.5 MULTIPLE-ACCESS PARAMETERS FOR EQ. (5.2.15)

MODE OF ACCESS	PARAMETER ASSIGNMENTS	
	P_{VA}	P_{BO}
FDMA	0 dB	3 dB
TDMA	0	0
FDMA-DA	0	3
TDMA-DA	0	0
FDMA-SPADE	4	3
ALOHAs	4	0
SLOHAs	4	0

TABLE 5.2.6 SATELLITE PARAMETERS FOR EQ. (5.2.15)

SATELLITE CONFIGURATION	BW	EIRP	L_s
WESTAR	36 MHz	33 dBw	196 dB
DSCS II	125	28	202
INTELSAT V	240	36	204.5

TABLE 5.2.2 SYSTEM IMPACT OF TDMA

ITEM	IMPACT
Transponder Power	i) Allows full use of power without backoff ii) When bandwidth limited, allows efficient bandwidth-power tradeoff to increase information rate using high-order modulation.
Transponder Bandwidth	When power limited, allows 2-5 dB FEC coding gain.
Frequency Plan	Greatly simplifies planning as all accesses use same frequency.
Terrestrial Network	Compatible with terrestrial digital network.
Reconfiguration	Allows easy adjustment of the capacity of each access.
Traffic Loading	i) Efficient for medium-high loading ii) Inefficient for light traffic loading.

modulation involved, but an example based on INTELSAT IV data is available and is shown in Figure 5.2.3. This figure shows clearly the huge penalty paid in capacity for multiple accesses of this system.

To improve the multiple-access capability of FDMA, a single channel per carrier (SCPS)-FDMA called SPADE has been deployed by INTELSAT. SPADE provides a pool of 800 channels which are shared by all earth stations in common view of the satellite. Furthermore, to conserve transponder power and to randomize carrier power occupancy throughout the transponder bandwidth (to alleviate the intermodulation problem), each carrier is activated or deactivated according to the activity of user speech. This leads to the following simplified system equation which can be used for channel capacity calculations (note that demand assignment is not included in this equation).

TABLE 5.2.3 FDMA TYPES

FDMA TYPE	STATUS	ADVANTAGES
FDM-FM-FDMA	In use	Efficient and inexpensive for very limited access trunk operation
SCPC-FDMA (SPADE)	Introduced	SCPC, efficient for large number of accesses with small users
PCM-PSK-FDMA	Under development	Allow substantial increase in channel capacity using DSI

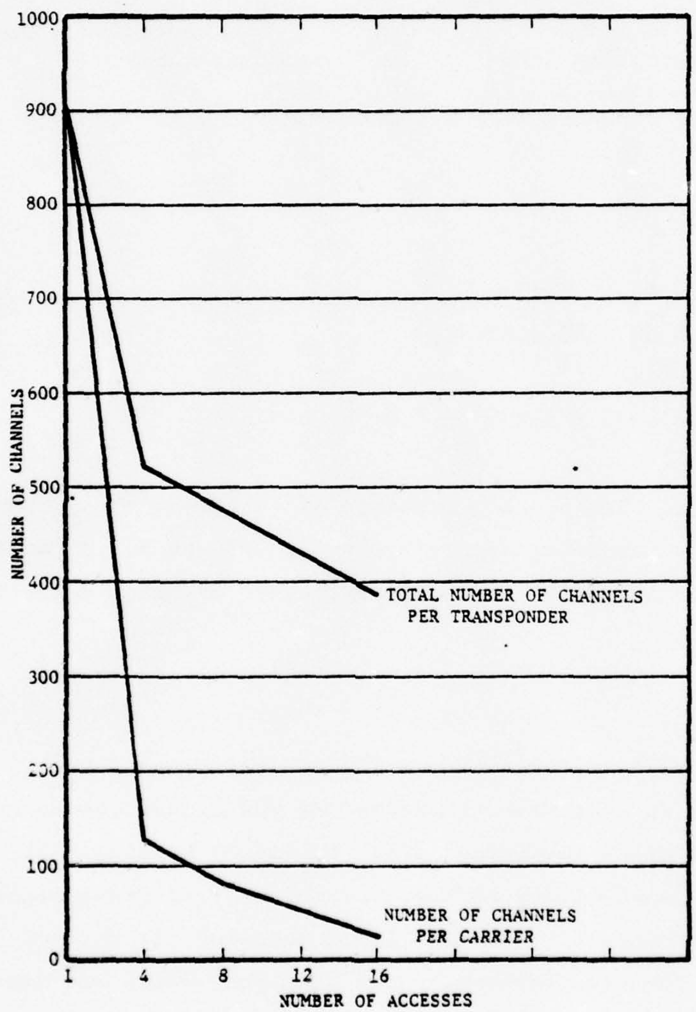


FIGURE 5.2.3 CHANNEL CAPACITY OF INTELSAT IV FM-FDM-FDMA (BASED ON COMSAT DATA)

$$m = EIRP + G/T - \frac{E_b}{N_o} - R - \text{margin} - L_s - k - P_{BO} + P_{VA} \qquad (5.2.4)$$

channel data rate
threshold
ground station G/T
satellite EIRP
channel capacity (number of channels @ R bps/channel)
voice activation
transponder backoff
Boltzman's constant
space loss

Assuming that

L_s = 197 dB (4 GHz)
P_{VA} = 4 dB (40% activity)
margin = 6 dB
E_b/N_o = 8.4 dB @ BER = 10^{-4} with QPSK

Eq. (5.2.4) can be plotted as a function of (EIRP + G/T) as shown in Figure 5.2.4. By comparing Figure 5.2.4 with Figure 5.2.3, we find that multiple-access capability is greatly increased by using SPADE type FDMA.

5.2.2.3 CDMA

A code-division multiple-access (DCMA) system makes use of approximately orthogonal codes so that many transmissions can occupy the same spectrum simultaneously. Other variations are: frequency hopping (FH), time hopping, or any combinations of these techniques. A pseudorandom noise (PN) code is the basis for the CDMA implementation; it spreads user signals into a wideband, low-power-density signal which has statistical properties somewhat similar to random noise. Because of this noise-like nature, it is quite tolerant of interference. As a result, the total number of users can exceed the maximum number of active users, thus providing an inherent DAMA capability. CDMA also provides message

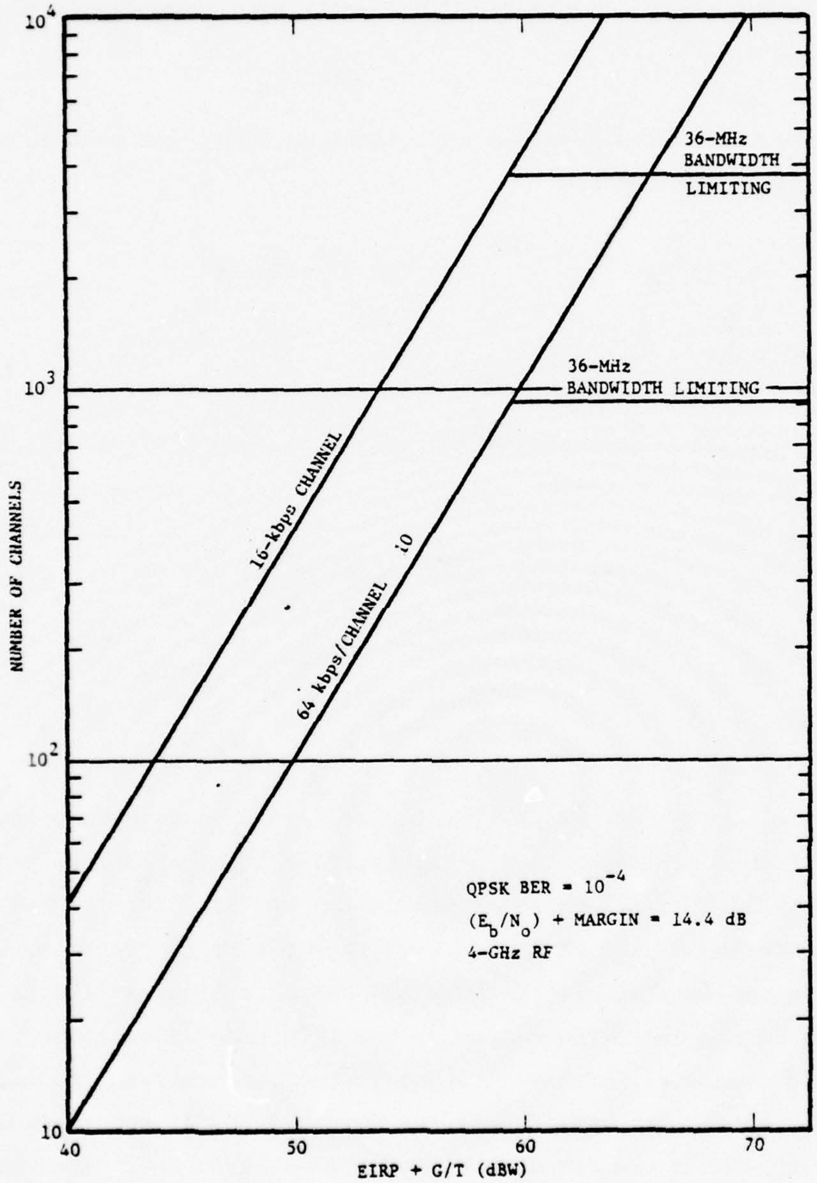


FIGURE 5.2.4 SCPC-FDMA CHANNEL CAPACITY (SPEECH ACTIVATED)

privacy, selective calling, random addressing (transmitted code serves as identification code) capabilities, and anti-jamming capabilities in general.

To assess the multiple-access capability of CDMA, let each access transmit a signal denoted by

$$\underbrace{a_i(t)}_{\text{code}}\ \underbrace{S_i(t)}_{\text{signal}} \qquad \text{for the } i^{th} \text{ access}$$

At the receiver, after despreading the signal by code correlation, the signal becomes:

$$\underbrace{\overline{a_i^2(t)}\ S_i(t)}_{\text{wanted signal}} + \underbrace{\sum_{\substack{j=1 \\ j\neq 1}}^{m} \overline{a_i(t)\ a_j(t)}\ S_j(t)}_{\text{unwanted portion}} \tag{5.2.6}$$

For theoretical orthogonal codes, the second term is zero; thus, the number of multiple access, m, can be arbitrarily high as long as there is enough satellite bandwidth and power to support it. This is not the case since the second term of Eq. (5.2.6) is not zero in practice. This non-zero term consumes satellite power and presents a noise-like interference (in addition to the usual thermal noise and intermodulation noise) to the receiving downlink, further limiting the multiple-access capability. However, this noise-like spread-spectrum interference is substantially rejected because of a concept of processing gain inherent to the system. This is illustrated in Figure 5.2.5.

The processing gain, g, is defined as the ratio of spread-spectrum PN code rate to information transmissions, i.e.,

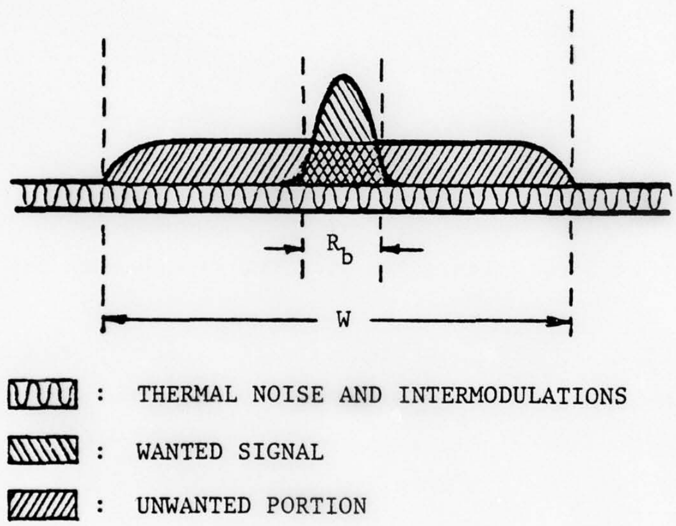


FIGURE 5.2.5 CONCEPT OF CODE DIVISION MULTIPLE ACCESS (CDMA)

$$g = \frac{P}{R_b} \qquad (5.2.7)$$

Since the second term of Eq. (5.2.6) is noise-like and is assumed to spread evenly across the entire band, W, the actual contributing spread-spectrum interference noise is only a fraction, 1/g, of the total. For ease of calculation, let the signal power be equal for all accesses; then, the signal-to-noise-power ratio for the i^{th} access after despreading can be shown as:

$$\frac{E_b}{N_o} = \frac{S_i}{(m-1)\ S_i \frac{1}{g} + (n_o + I_m) \frac{R_b}{W}} \qquad (5.2.8)$$

where n_o is the total noise power of thermal noise and
I_m is intermodulation noise over the transponder bandwidth, W.

Equation (5.2.8) is difficult to evaluate except in very specific cases because the intermodulation noise is dependent on the multiple-access number, m, in a nonlinear way. By assuming $I_m = 0$, the expression is greatly simplified; in this case, we have:

$$\frac{E_b}{N_o} = g \frac{(S/N)_s}{(m-1)(S/N)_s + 1} \qquad (5.2.9)$$

where R_b/W is assumed to equal 1/g for simplicity and
$(S/N)_s$ is the spread-spectrum signal-to-noise power ratio at the receiver input.

Solving Eq. (5.2.9) for m, we have:

$$m = \frac{g}{E_b/N_o} + \left[1 - \frac{1}{(S/N)_s}\right] \qquad (5.2.10)$$

which is plotted in Figure 5.2.6.

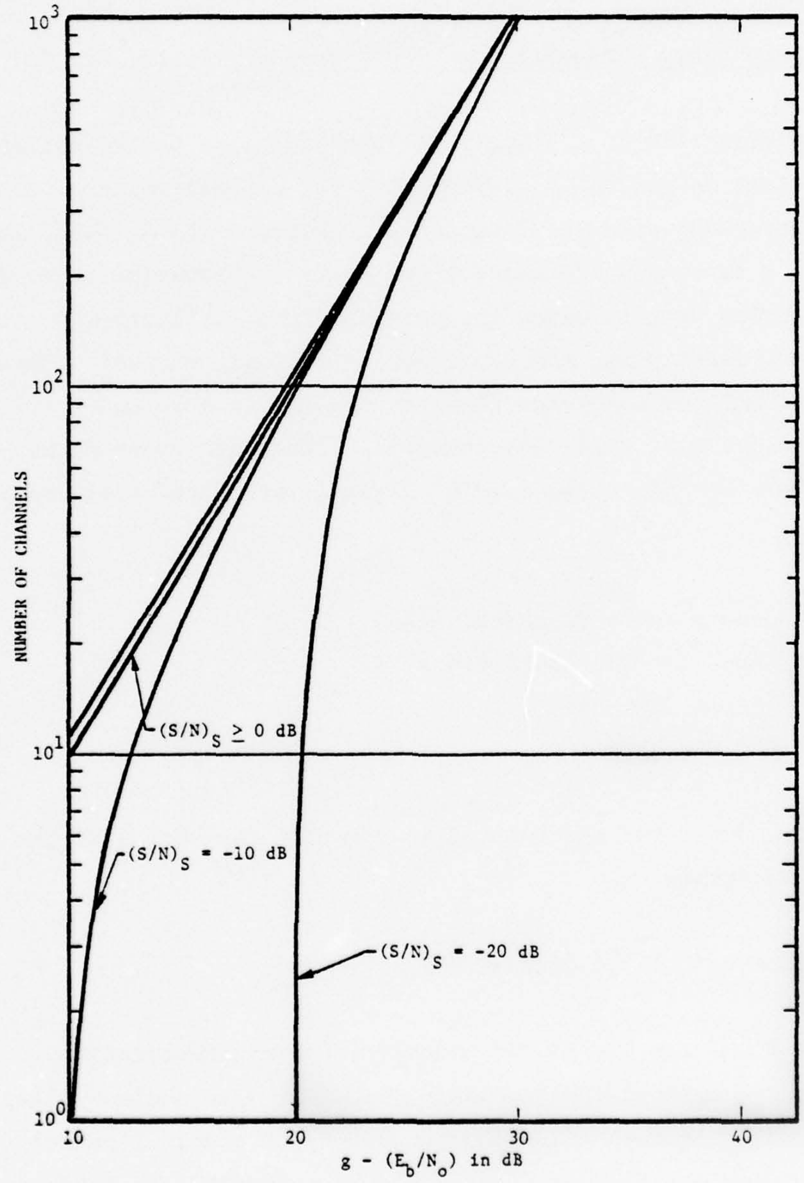


FIGURE 5.2.6 MULTIPLE-ACCES CAPABILITY OF CDMA

A comparison of several CDMA techniques is given in Table 5.2.4.

5.2.3 Packet-Switched Channelization Techniques [2, 8, 17, 22, 28]

The channelization of the available bandwidth for packet-switched satellite systems is generally simpler than for circuit-switched DAMA systems. In contrast to circuit switched techniques (FAMA, DVDA, FVDA) which provide a two-way voice channel for every conversation in progress, a packetized voice system, using for example ALOHA, utilizes the entire satellite transponder power and bandwidth as a single channel. Some reservation techniques subdivide the main channel into an orderwire channel and one or more time-slot channels. These different packet-switched systems can be categorized by channelization requirements as follows:

- One Channel - No Synchronization
- One Channel - Synchronization
- Subslotted Channels
- CAPTURE Channels

These cases are discussed and related to specific techniques in the following subsections.

5.2.3.1 One Unsynchronized Channel

For pure ALOHA the ETs do not maintain time-synchronization. Packet bursts are emitted upon assembly (or after a randomized delay for retransmissions). As with a TDMA burst however, the actual packet (header plus information) must be preceded by a preamble of overhead bits for carrier-recovery (CR) and bit timing recovery (CR/BTR). As discussed in Section 5.2.2.1 for TDMA the current PLL state-of-the-art requires a CR/BTR sequence of about 100 bits. To this overhead a 20 bit unique word (UW) must also be added to establish the start of the actual packet.

TABLE 5.2.4 COMPARISON OF SEVERAL SPREAD-SPECTRUM TECHNIQUES

TYPE	TECHNIQUE	ADVANTAGES	DISADVANTAGES
PN	PSEUDONOISE CODE ⊕ PSK DATA CARRIER	1. SIMPLE RF EQUIPMENT 2. COMPATIBLE WITH COHERENT DATA MODULATION - BEST POWER EFFICIENCY	1. DIFFICULT CODE ACQUISITION • COMPLEX EQUIPMENT • TIME CONSUMING
FH	FREQUENCY HOP ⊕ MFSK DATA CARRIER	1. SIMPLER CODE ACQUISITION • LESS TIME • LESS COMPLEX 2. NO CARRIER ACQUISITION • LESS TIME TO ACQUIRE	1. COMPLEX RF GENERATION EQUIPMENT (FREQUENCY SYNTHESIZER) 2. LESS POWER EFFICIENT
TH	TIME HOP ⊕ PSK OR DPSK DATA CARRIER	1. COMPATIBLE WITH PACKET XMISSION TECHNIQUE	1. REQUIRES HIGH PEAK POWER XMITTER AT EARTH TERMINAL
HYBRID	COMBINATION OF ANY TWO ABOVE	1. FLEXIBILITY IN OPTIMIZATION FOR GIVEN PROCESSING GAIN	1. GREATER COMPLEXITY, COST

The burst rate requirements as determined for ALOHA in Chapter 4 is only for information and abbreviated header bits. The actual burst rate is related to that information burst rate by the packet length expansion of these overhead bits:

$$\frac{BR}{C} = \frac{\text{ACTUAL BR}}{\text{INFO BR}} = \frac{120 + \text{PACKET LENGTH}}{\text{PACKET LENGTH}}$$

Specializing this to 20 ms voice packets for 16 kbps CVSD (335 bits) and 4 kbps PEV (95 bits) gives ratios of

$$(16 \text{ kbps CVSD}) : \frac{\text{ACTUAL BR}}{\text{INFO BR}} = 1.36$$

$$(4 \text{ kbps PEV}) : \frac{\text{ACTUAL BR}}{\text{INFO BR}} = 2.32$$

5.2.3.2 One Synchronized Channel

For slotted ALOHA, the ETs synchronize their times of possible packet transmission. This synchronization requires no more special bits besides the CR/BTR and UW (120 bits) than for the unsynchronized channel. To effect the required synchronization the ETs can operate a slot-synchronizing loop which adjusts its timing by comparing the beginning of its own packets and the packets of any ET designated as the master clock.

While no additional bits are needed to effect slot-synchronization, a rule-of-thumb guard time of about 200 ns must be placed between packet bursts to ensure no overlap because of slight timing errors. The actual burst rate must again be increased to achieve the required information rate over the total slot time. The previous unslotted BR/C formula then becomes:

$$\frac{BR}{C} = \frac{120 + \text{INFO BITS}}{(\text{INFO BITS}) - \text{Cx } (\text{GUARD TIME})}$$

If 200 ns is the actual state-of-the-art, then BR explosively increases as C approaches 1675 Mbps for 335 bit packets and 475 Mbps for 95 bit packets. Since these rates are in the range required for the slotted ALOHA baseline system, slotted ALOHA may be unusable. Since this issue is quite sensitive to guard time, the guard time issue should be investigated more closely. The accompanying analysis (Chapter 4) ignores this problem pending a definitive answer.

5.2.3.3 Subslotted Channels

Reservation schemes (e.g., Round Robin and Robert's Reservation) require a slot synchronization plus one or a number of sub-slots for reservations and data.

Since the subslots are accessed by different ETs, guard times are required between sub-slot intervals and CR/BR plus UW bits are required between every sub-slot. The sub-slots can then be considered equivalent to the synchronized slot problem described in the previous section.

5.2.3.4 CAPTURE Channels

For a CAPTURE processing satellite operated with slotted ALOHA protocol, every up-channel is distinct (orthogonal), therefore small timing errors in the packet transmission of one ET cannot cause errors in the packets transmitted by another ET. Thus, no guard time would seem to be required. Although not analyzed at this time, a CAPTURE system should be operable in an unsynchronized mode. If either of these assumptions proves to be correct, then the unsynchronized channel (Section 5.2.3.1) overhead results apply:

$$\frac{BR}{C} = \frac{\text{ACTUAL BR}}{\text{INFO BR}} = \frac{120 + \text{PACKET LENGTH}}{\text{PACKET LENGTH}}$$

Specializing this to 20 ms voice packets for 16 kbps CVSD (335 bits) and 4 kpbs PEV (95 bits) gives ratios of:

$$(16 \text{ kbps CVSD}): \quad \frac{\text{ACTUAL BR}}{\text{INFO BR}} = 1.36$$

$$(4 \text{ kbps PEV}): \quad \frac{\text{ACTUAL BR}}{\text{INFO BR}} = 2.32$$

5.2.4 Voice Channel Capacity of MA Techniques for Reference Satellites

To assess the channel capacity of the multiple-access techniques, calculations are made by expressing the number of 16-kbps voice channels as a function of earth terminal G/T for the following access modes and satellite configurations:

ACCESS MODES

- FDMA
- TDMA
- FDMA-DA
- TDMA-DA
- FDMA-SPADE
- ALOHA without retransmission @ P_L = 25%
- ALOHA without retransmission @ P_L = 10%
- SLOHA without retransmission @ P_L = 25%
- SLOHA without retransmission @ P_L = 10%

SATELLITE CONFIGURATIONS

1. WESTAR
   C-Band
   BW = 36 MHz
   EIRP = 33 dBW
2. DSCS II
   X-Band
   BW = 125 MHz
   EIRP = 25 dBW
3. *INTELSAT V
   K_u-Band
   BW = 240 MHz
   EIRP = 36 dBW

---

*INTELSAT V uses spot beams at K_u-band with EIRP = 47 dBW, but is defocused to earth coverage for calculations here. Frequency reuse is not included.

Calculations are based on the following equation in dB.

$$G/T = (E_b/N_o) + R_b + M + \text{margin} - P_{VA} + P_{BO} + L_s + k - \text{EIRP} \quad (5.2.31)$$

satellite EIRP

Boltzman's constant

free space loss

3 dB transponder backoff, if any

voice activation, if any

6 dB

M = 10 log(N), N = # of 16 kHz channels

16 kbps

= 7 dB for BER = 10^{-3} with QPSK

Note that this equation does not take into account miscellaneous losses which are not mode-common. Also, it does not differentiate between TDMA and FDMA except a 3 dB backoff (which is not optimized for intermodulation products).

The assignments of parameters in Eq. (5.2.15) for each access mode and each satellite configuration are listed in Tables 5.2.5 and 5.2.6, respectively. The results are tabulated in Table 5.2.7 and are plotted in Figures 5.2.7 through 5.2.9. The results show that, among those considered, the FDMA-SPADE needs the least amount of G/T to reach the capacity limit of the satellite. Note also that TDMA behaves similar to SLOHA at P_L = 10%, link-wise, but the significant difference is that the number of earth terminals limits the number of users (illustrated in Figure 5.2.3), while SLOHA does not have such a limitation.

TABLE 5.2.5 MULTIPLE-ACCESS PARAMETERS FOR EQ. (5.2.15)

MODE OF ACCESS	PARAMETER ASSIGNMENTS	
	P_{VA}	P_{BO}
FDMA	0 dB	3 dB
TDMA	0	0
FDMA-DA	0	3
TDMA-DA	0	0
FDMA-SPADE	4	3
ALOHAs	4	0
SLOHAs	4	0

TABLE 5.2.6 SATELLITE PARAMETERS FOR EQ. (5.2.15)

SATELLITE CONFIGURATION	BW	EIRP	L_s
WESTAR	36 MHz	33 dBw	196 dB
DSCS II	125	28	202
INTELSAT V	240	36	204.5

TABLE 5.2.7 CHANNEL CAPACITY OF MULTIPLE-ACCESS TECHNIQUES

MULTIPLE-ACCESS MODE	NUMBER OF 16-kbps CHANNEL, M, AS FUNCTION OF G/T (EXPRESSED IN dB)		
	WESTAR	DSCS II	INTELSAT V
FDMA	M = G/T + 7.56	M = G/T = 3.44	M - G/T + 2.05
TDMA	M = G/T + 10.56	M = G/T - 0.44	M = G/T + 5.05
FDMA-SPADE	M = G/T + 11.56	M = G/T + 0.56	M = G/T + 6.05
ALOHA without retransmission @ P_L = 25%	M = G/T + 6.02	M = G/T - 4.98	M = G/T - 0.95
ALOHA without retransmission @ P_L = 10%	M = G/T + 1.55	M = G/T - 9.45	M = G/T - 3.96
SLOHA without retransmission @ P_L = 25%	M = G/T + 9.18	M = G/T - 1.82	M = G/T + 3.67
SLOHA without retransmission @ P_L = 10%	M = G/T + 4.55	M = G/T - 6.45	M = G/T + 0.51

5.3 SPACE SEGMENT REALIZATION CONCEPTS

5.3.1 Transponder Utilization

Efficient use of transponder power is important in a DAMA environment. There are a number of ways in achieving this:

i) Non-contentional power sharing
ii) Contentional power sharing
iii) More powerful transponder development

Non-Contentional Power Sharing (NCPS)

This refers to the conventional power-sharing concepts

- Frequency Division : FDMA
- Time Division : TDMA
- Code Division : CDMA
- Space Division : spot-beam antenna
- Polarization Division : polarization diversity

Contentional Power Sharing (CPS)

This concept refers to the idea of random power sharing based on the statistical unlikelihood of simultaneous demand for power among a collection of diverse users:

- ALOHA and variations
- SPADE - compatible with NCPS
- DSI - compatible with NCPS

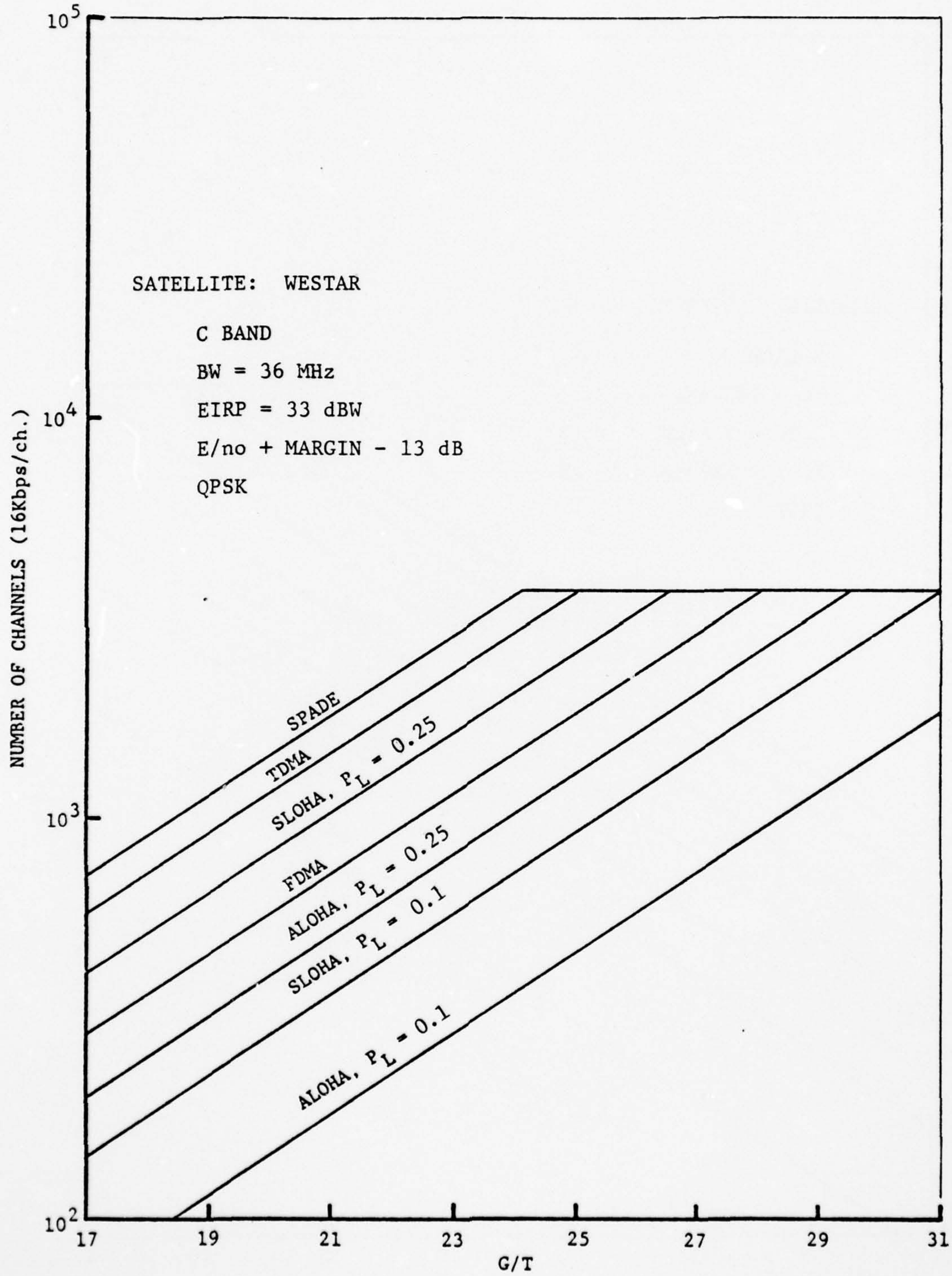


FIGURE 5.2.7 RESULTS OF EQ. (5.2.15) FOR WESTAR (C-BAND)

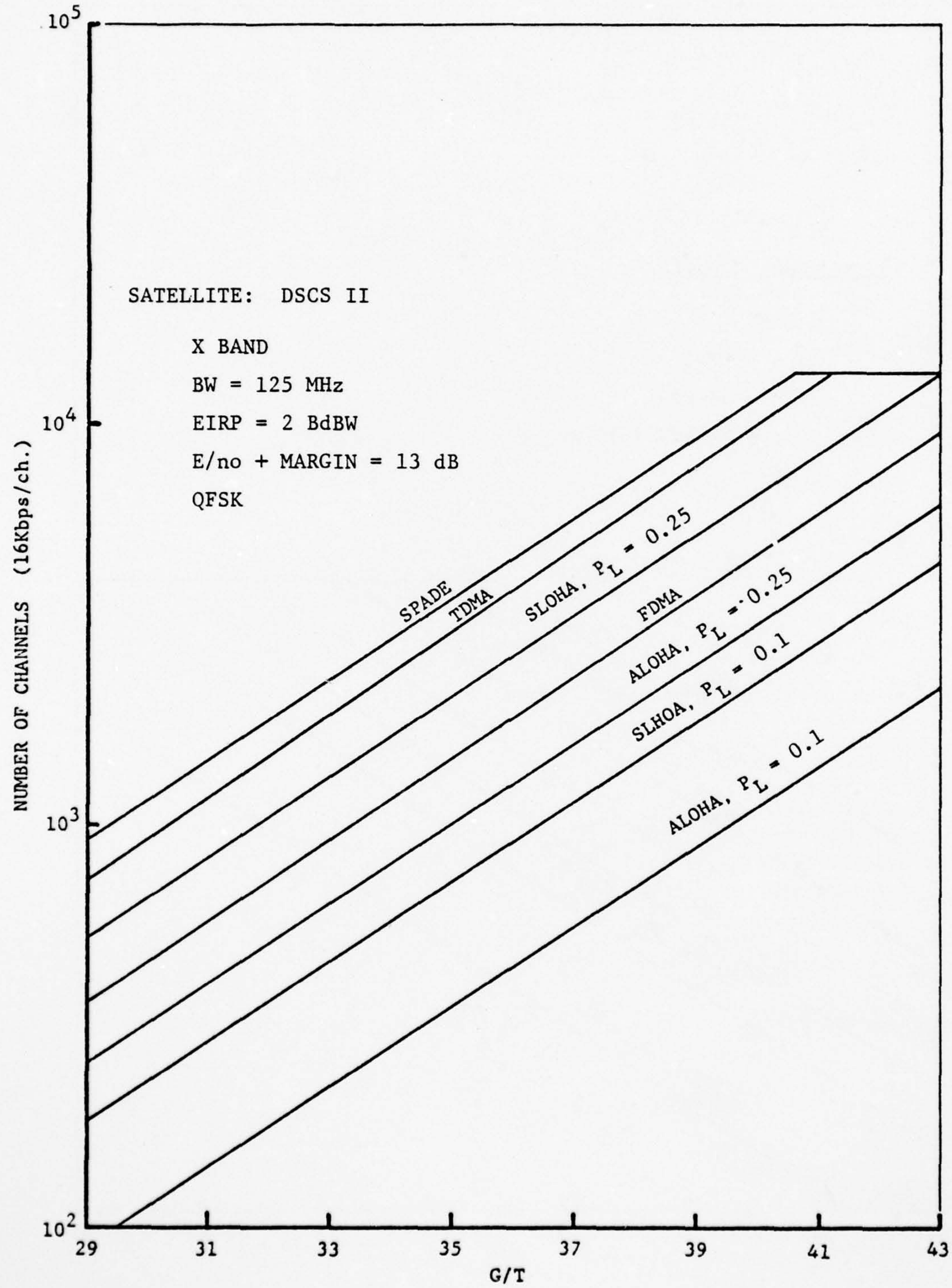


FIGURE 5.2.8 RESULTS OF EQ. (5.2.15) FOR DCS III (X-BAND)

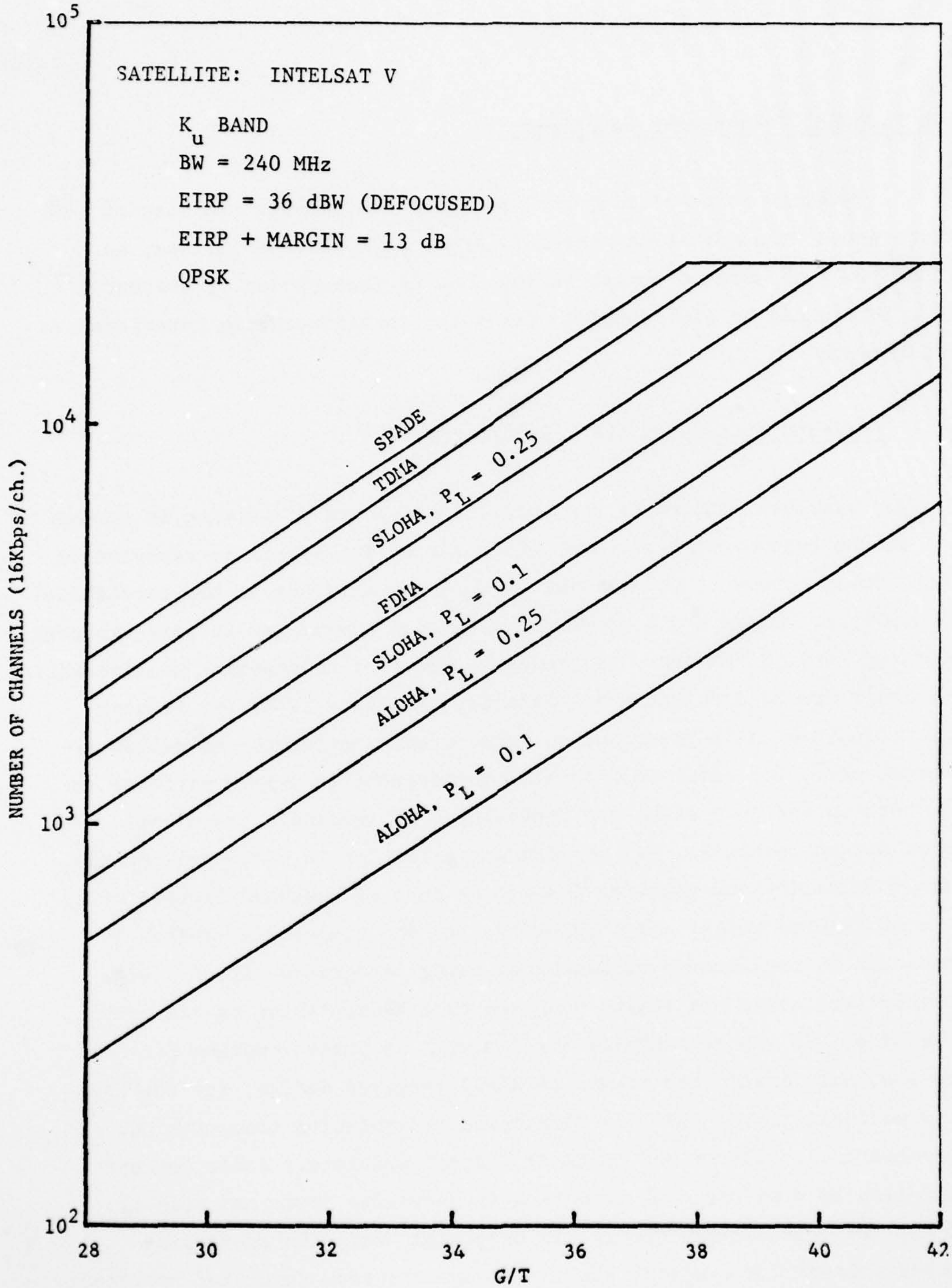


FIGURE 5.2.9 RESULTS OF EQ. (5.2.15) FOR INTELSAT V (K_u-BAND)

More Powerful Transponder Development

In addition to developing more powerful transponders in view of future space transportation systems' larger payload capabilities, an extra 1 to 4 dB more power can be realized if transponder of present power levels can be linearized to relax the intermodulation interferences requirements.

5.3.2 On-Board Processing [14, 18, 42, 68]

The simplest concept of the satellite on-board processing is perhaps that of the regenerative repeater where the uplink signal is rejuvenated before being converted to downlink in much the same way as the terrestrial repeaters do. There are a number of potential advantages to this approach according to R. S. Davies: (a) reduces impact of transponder nonlinearities, especially for bandlimited QPSK operating through a TWTA, (b) reduces degradation caused by polarization interference, adjacent-channel interference, etc., (c) leads to 2 to 3 dB improvement in system performance when both up and down links are operating near threshold (as might happen during rainfall), (d) if coherent detection is used, much of the channel filtering and equalization can be done at baseband instead of at RF, with reduced weight and complexity, (e) the transponder-switched matrix can be implemented at baseband, using integrated circuits with possibly less power and weight compared to a RF-switching matrix using PIN-diodes, (f) the remodulated carrier will be phase coherent for all accesses, simplifying the ground terminal receiver design, (g) the need for a multiplicity of TWTAs is eliminated by replacing them with one high-powered oscillator and multiple digital modulators (this concept would lead to a significant reduction in satellite power and weight), and (h) the regenerative concept is compatible with the store-and-forward concept for use with the single antenna beam described previously. The store-and-forward concept also allows a shift in data transmission rate to take place in the satellite, a potential advantage when large and small terminals are included in the same network.

A recent Ford Aerospace & Communications Corporation (formerly Aeronutronic Ford Corporation) WDL study analyzing the performances of regenerative repeaters using QPSK and MSK shows that the regenerative repeater yields an improvement of 2 to 6 dB over the conventional translating repeater, depending on link conditions and inter-symbol interference level. A 4-dB improvement was reported in the 1976 IEEE Canadian Conference on Communication and Power through an experimental study of a similar problem.

Another concept of on-board processing is the satellite-switched multiple beams using Ka-band communication satellite systems. The major disadvantage of Ka-band, rainfall attenuation, can be overcome by increased link margin and by the use of more than one ground terminal (space diversity). An advantage of Ka-band is that narrow antenna beamwidths can be generated with reasonable size antennas on the satellite and ground. Narrow beamwidths permit reduced satellite spacing, thus conserving orbit space, and frequency reuse, which conserves frequency spectrum.

The basic system concept is shown in Figure 5.3.1. The satellite employs a multiple beam antenna which illuminates, either simultaneously or sequentially, zones on the earth where earth terminals are located. Multiple transponders are required if more than one antenna beam is used at a time, and a switching matrix to interconnect the signals between the various beams.

On-board processing, thus, opens up the possibility and potential advantages of the DAMA concept through multiple transponders.

5.3.3 Satellite Antenna

Satellite antennas can generally be grouped as:

- Earth coverage

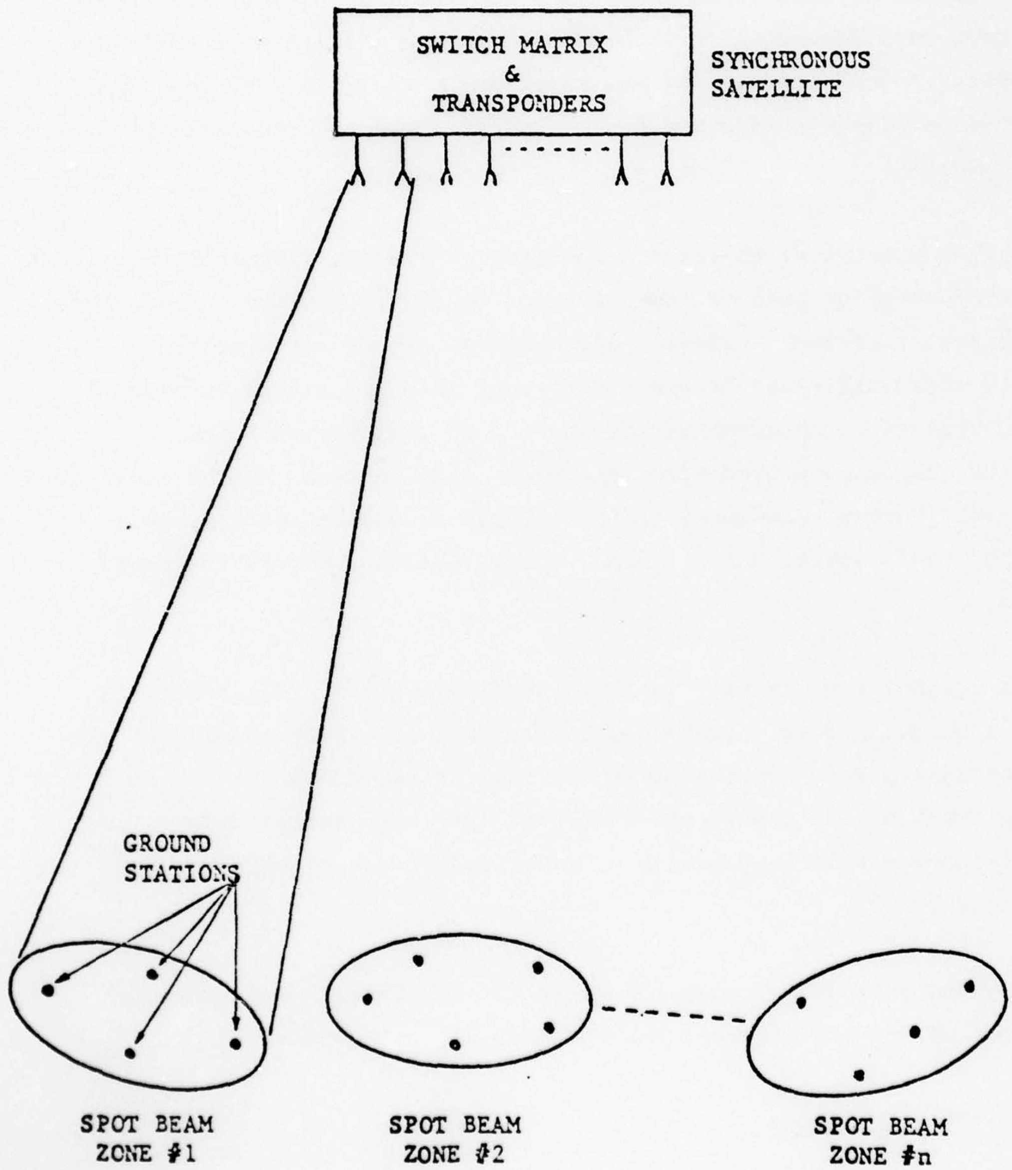
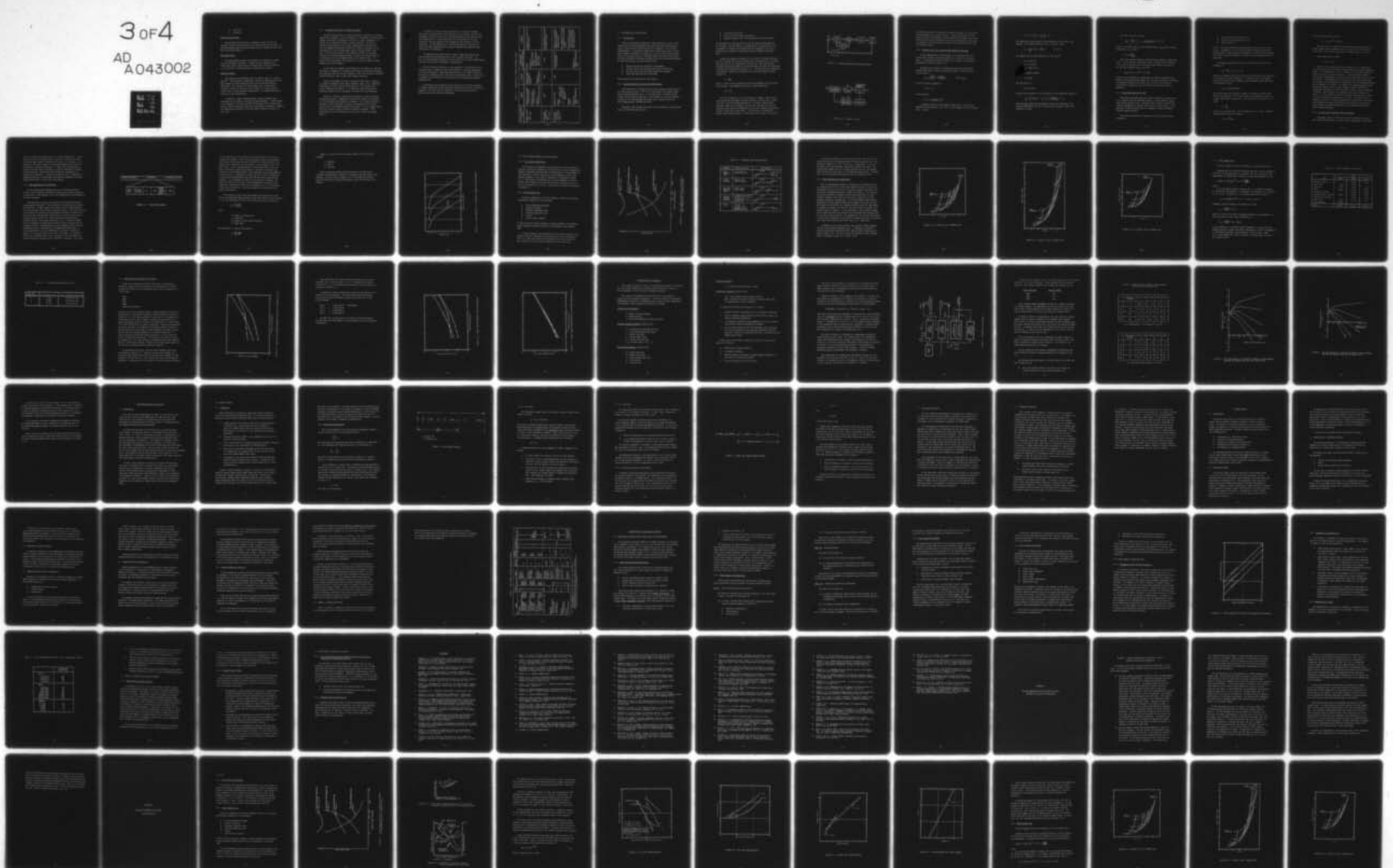


FIGURE 5.3.1 SYSTEM CONCEPT OF MULTIPLE-BEAM ANTENNAS

AD-A043 002 SYSTEMS CONTROL INC PALO ALTO CALIF F/G 17/2
IMPLICATIONS OF DEMAND ASSIGNMENT FOR FUTURE SATELLITE COMMUNIC--ETC(U)
JUN 77 C G HILBORN, A C PAN, P J BOGERT DCA100-76-C-0060

UNCLASSIFIED NL

3 OF 4
AD A043002



- Spot beam
- Hemi/zone

Earth-Coverage Antenna

Earth-coverage antennas have a beamwidth of about 18° and are intended to serve entire earth areas as observed from the satellite. It provides maximum interconnectivity among users but, generally, has lower gain (16.8 dB for DSCS II).

Spot-Beam Antenna

Spot-beam antennae usually are designed to be multiple in number and are individually steerable. Because of narrow beamwidth (2.5° covers about 1000 x 1000 nm area) and higher gain (33 dB for DSCS II), it is efficient in providing service to high-traffic regions.

Hemi-Zone Antenna

Zone antennas have beamwidths that are greater than 2.5° and less than 18°. They represent a compromise in terms of interconnectivity and gain between the earth-coverage antenna and the spot-beam antenna. Furthermore, the footprints of zone antennas can be controlled by exciting a cluster of feed horns which illuminate a large reflector (offset feed to avoid excessive interference). The shaped beams can be tailored to user distributions.

In addition to these classifications, various frequency reuse techniques can be applied to properly designed antennae. A prime example is the INTELSAT V to be built by WDL of Ford where, for the first time, both the spatial frequency reuse and the polarization diversity techniques are simultaneously applied to the same antenna (hemi-zone antenna, in this case).

5.3.4 Vulnerability Aspects of Demand Assignment

Another important aspect of demand assignment, especially in military communication systems, is its vulnerability to intentional or unintentional jamming or spoofing, interference, failure of a control component, or failure of an operator to follow a correct procedure. We will not consider the vulnerability of the data communications link directly (such as jamming the satellite uplink) but, rather, concentrate on the vulnerability of the DA system which controls the data links. Note that, if the DA system relies on an orderwire circuit for successful operation, an optimum jamming strategy to disrupt the network might be to concentrate all of the jamming power on the orderwire link. It appears that the orderwire circuit shall be at least as immune to enemy jamming as the data channels.

If the enemy can simulate user terminals in the system, an effective alternative to jamming is to saturate the system with simulated terminals; thus, denying access to the user terminals. Thus, the DA system, including orderwire circuits, must be made secure against spoofing.

Failure of equipment or operator procedure could cause a terminal to transmit erroneously (wrong time, frequency, power, etc.), generating interference to other users in the network, or, in an extreme case, totally disabling the network. It is important that the selected DA technique contain built-in safeguards to prevent this from happening.

Table 5.3.1 presents a qualitative evaluation of three DA systems discussed previously. Examination of this table reveals the following: (1) All DA systems are vulnerable to jamming. (2) A central-control DA technique is more vulnerable to failure of equipment (at the control station) than a distributed-control DA technique. (3) All systems require some form of secure coding to prevent takeover by spoofing. (This coding might be the spread-spectrum code used to counter the jamming threat.)

In general, the more reliance (expressed as a required orderwire data rate) a system places on the existence of an orderwire, the less the throughput efficiency will be for a given satellite, ground terminal size, and jamming threat. On the other hand, if the orderwire is too well protected, then the enemy will find jamming the data channel to be more profitable. In the limiting case, the secured-orderwire circuits also become the data channels with all available terminal and satellite power allocated to the orderwire.

An orderwire using CDMA instead of TDMA or FDMA would already have some spread-spectrum protection built into the signal structure. In fact, a coded orderwire modulation could be devised to satisfy channelization, anti-jam, and security requirements, simultaneously.

Another evaluation criteria, not yet mentioned, is the ability of the network to make a transition from an unstressed to a stressed environment. A DA system employing a spread-spectrum orderwire at all times, obviously, can make this transition more easily than a system which must, at the outset of jamming, switch modulation, acquire codes, and re-establish orderwire communications.

A DA system using processing in the satellite will be less vulnerable to jamming (especially uplink) than those using simple frequency translation repeaters. The extra cost and complexity of these satellites must be balanced against this improved performance.

TABLE 5.3.1 VULNERABILITY OF DEMAND-ASSIGNMENT SYSTEMS

DEMAND-ASSIGNMENT SYSTEM	JAMMING	SPOOFING	INTERFERENCE (ERRONEOUS XMISSION)	COMMENTS
CENTRAL-POLLED TDMA (OW DATA)	JAMMING OF OW UPLINK: DISABLES NETWORK DOWNLINK (LOCAL): DISABLES TERMINAL IMPROVE BY: SPREAD SPECTRUM ON OW	REDUCES CAPACITY BY CAPTURING CHANNELS IMPROVE BY: CRYPTO CODING	TIMING IMPROVE BY: -SPREAD SPECTRUM ON DATA -INHIBIT XMISSION WHEN LOSS OF TIMING SYNC	VULNERABLE TO FAILURE OF CENTRAL CONTROL
DISTRIBUTED-INTERRUPT OW TDMA (PW DATA)	SAME	SAME	SAME	NOT VULNERABLE TO FAILURE OF CENTRAL CONTROL
DISTRIBUTED-INTERRUPT DATA - TDMA (PACKET)	JAMMING OF DATA: UPLINK: PULSE - PARTIAL LOSS OF CAPACITY CW - TOTAL LOSS OF CAPACITY DOWNLINK: SAME IMPROVE BY: SPREAD SPECTRUM ON DATA	SAME	LESS SUSCEPTIBLE TO TIMING ERRORS IMPROVE BY: SPREAD SPECTRUM ON DATA	JAMMING OF DATA PREVENTS TERMINAL TO MONITOR NETWORK STATUS NOT VULNERABLE TO FAILURE OF ORDERWIRE OR CENTRAL CONTROL SPREAD SPECTRUM A/J GAIN LESS ON DATA CHANNEL WHEN DATA RATE GREATER THAN ORDERWIRE RATE

5.4 IMPLEMENTATION CONSIDERATIONS

5.4.1 Introduction

Given the conceptual feasibility of RMA deployment in terms of the required satellite bandwidth/EIRP allocations and earth terminal sizes (G/T), other technical limitations must be considered in the eventual implementation. An exhaustive exploration of every major area of potential technical limitation as per RMA is out of the scope of this section. Rather, this section serves to highlight some of the areas which should be further examined to determine technical risks and existing developments. Of particular concern are the following items:

- Spread Spectrum code acquisition requirements
- Low duty cycle, fast acquiring AGC receiver technology
- Low duty cycle, high peak power amplifier technology
- Guard time reduction for slotted RMA

These items are each addressed in this section.

5.4.2 Spread Spectrum Code Acquisition Requirements

Code acquisition is necessary in all spread spectrum systems because the spread spectrum code is the key for despreading the intended signal while spreading undesired ones. In the application of spread spectrum to RMA, code acquisition time must be short enough so that the resulting overhead is not overburdening the packet burst, typically short for voice communications (300-400 data bits).

In general, code acquisition times are lower bounded by uncertainties due to the following main sources:

- Code phase uncertainty
- Code clock and carrier uncertainties
- Doppler frequency shift and propagation time uncertainties

The acquisition of code involves the steps of initial synchronization and tracking. Many techniques of varying degree of sophistication can be employed for initial code synchronization, depending on the type of application, the amount of uncertainty and the allowable acquisition time.

Perhaps the simplest technique is the so-called "Sliding Correlator." For this technique the receiver operates its code generator at a different rate than that of the transmitters, resulting in a two code sequence slipping in phase with respect to each other until the point of coincidence is reached as illustrated in Figure 5.4.1. To assess the code acquisition time required, assume that the general rise time-to-bandwidth relation is given by

$$T_c = \frac{C}{BW}$$

where C is some circuit constant and BW is the bandwidth of post correlation receiver. The maximum search rate is them approximately

$$R_s = \frac{2}{T_c}$$

For BW = 10 MHz and C = 0.35, we have R_s = 57 Mbps. Thus in this case the maximum time needed to search through a 128 chip code is about 2.25 μsec, too long for RMA applications. In order to apply spread spectrum to RMA, it is possible to use only one 128 chip code for all multiple access since the probability that any two accesses transmitting exactly at same time is very small. In this case, it is possible to acquire code synchronization in a few bits or on the order of 35-50 μsec.

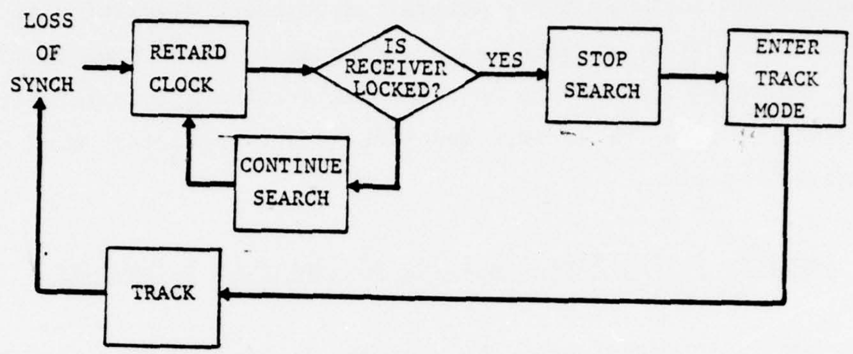


FIGURE 5.4.1 SPREAD SPECTRUM CODE ACQUISITION LOOP

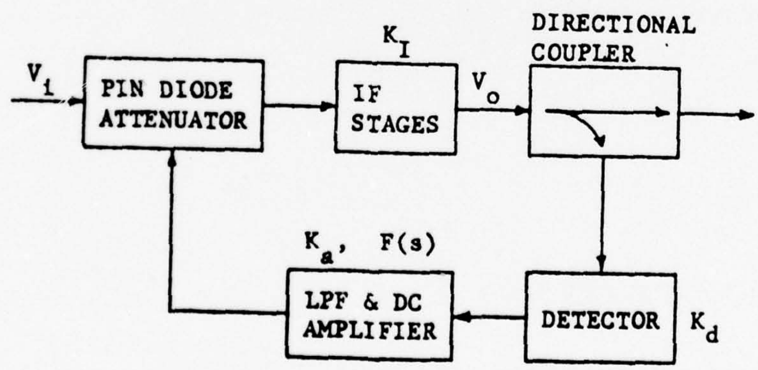


FIGURE 5.4.2 TYPICAL AGC LOOP

An acquisition time of such amount is perhaps acceptable for the traffic loading assumed in this study, however, a thorough discussion of this matter does not seem warranted as system analyses performed earlier have already indicated that SS-RMA does not yield any appreciable system gain as compared to pure RMA schemes and their other variations such as RMA with multiple copies.

5.4.3 Low Duty Cycle, Fast Acquiring AGC Receiver Technology

In the implementation of RMA systems, it is essential that the receiver AGC circuitry functions properly in a low duty cycle bursty mode. To illustrate the issue involved, let a typical AGC circuitry be represented as shown in Figure 5.4.2.

Assuming that the input is noiseless and that the AGC loop has zero-dB noise figure, the loop transfer function is, from the servo-control theory,

$$H(s) \triangleq \frac{V_o(s)}{V_i(s)} = \frac{K\,F(s)}{1 + K\,F(s)}\,, \qquad K \triangleq K_I\,K_d\,K_a$$

The AGC loop error, defined as

$$e \triangleq V_i - V_o$$

is then given by

$$e(s) = \frac{1}{1 + K\,F(s)}\,V_i(s)$$

A convenient measure of the dynamic range error is via the step response of the loop error, e(s), due to a step change, D, in the carrier level. That is,

$$T(s) = e(s).\frac{d}{s} = \frac{1}{1 + K\,F(s)} \cdot \frac{D}{s}$$

The steady-state step response is then obtained from the final value theorem. For a simple FC filter, $F(s) = 1/(1+\tau s)$. Then,

$$T_{\infty} = \lim_{s\to 0} s\,T(s) = \frac{D}{1 + K} \qquad \text{if } K \gg 1.$$

For example, let the AGC coefficients of the loop be

$$K_I = 5 \text{ dB/volt}$$

$$K_d = 14 \text{ mV/dB}$$

$$K_a = 1000 \text{ volt/V.}$$

for a dynamic range of

$$D = \pm 7 \text{ dB,}$$

the step error is:

$$T_{\infty} = \pm 0.1 \text{ dB.}$$

Another item of interest is the one-sided AGC loop bandwidth defined as:

$$B_{AGC} \triangleq \int_0^{\infty} |H(j\omega)|^2 \quad df = \int_0^{\infty} \left|\frac{K\,F(j\omega)}{1 + K\,F(j\omega)}\right|^2 \quad df$$

Note that B_{AGC} should be wide enough to allow fast following of the carrier level variations, yet it must be narrow enough to reduce the influence of noise.

For $F(s) = 1/(1+\tau s)$, we have

$$B_{AGC} = \frac{K}{k+1} \int_0^{\infty} 1 + \frac{1}{\{\omega/[(k+1)/2]\}^2} \, df = \frac{K}{4\tau}$$

Since $T_{\infty} = D/(1+K) \doteq D/K$, the relationship between B_{AGC} and the dynamic range error, T_{∞}, is then:

$$T_{\infty} = \frac{D}{4\tau \; B_{AGC}}$$

Note that the brief analysis presented above assumes a steady-state input. For an input of a RMA packet burst of length of about 600 bits or 4.7 μsec at 128 Mbps rate, the steady state results can be used if the GAC loop bandwidth B_{AGC} is such that

$$B_{AGC} \geq 2/4.7 \times 10^{-6} = 427 \text{ KHz},$$

which is not too difficult a requirement for hardware implementation. Moreover this bandwidth requirement can be further relaxed by introducing an integrate-and-dump filter memory into the AGC loop to obtain a burst-to-burst average so that the effect of the low duty cycle is minimized.

5.4.4 Guard Time Reduction For RMA

Transmission Systems using time division multiplex techniques such as the slotted ALOHA and TDMA/DVRA require a certain amount of guard time between adjacent slots. The guard time is needed to prevent the possible overlapping of channels due to errors in slot timing. It is important to keep the guard time to a minimum to insure the highest possible frame efficiency.

The guard time required is dependent upon the following circuit parameters:

- accrued error per correction (e_e)
- clock rate and stability (e_c & e_s)
- logic filter (e_j)

That is, the guard time must be large enough to account for all these errors. Furthermore, if the ranging uncertainty induced timing error (using whatever ranging technique applicable) is appreciable as compared with those of circuit parameters, it should also be accounted for in the guard time.

The minimum required guard time, T_g, between packet burst can be expressed as

$$T_g = 2[2e_e + e_c + e_s + e_j]$$

where e_e is being accounted for twice as the accrued error per correction can be either positive or negative. To digress further, the maximum correction rate C_M is upper bounded by the propagation round trip delay (0.25 seconds) so that

$$C_M \simeq 4 \text{ corrections/sec}$$

The accrued error per correction, which is a function of orbit eccentricity, and using CM = 4, is on the order of 1 nanosecond or less, not a major contributor to guard time. The error due to clock rate, e_c, is simply

$$e_c = \frac{1}{R_c}$$

where R_c is the clock rate. For a 100 MHz clock, e_c = 10ns. Assuming further that the clock is stable to

$$\pm 1 \times 10^{-9}/\text{day}$$

Then the error per 0.28 second is

$$e_s = .25 \times 10^{-9} = 0.25 \text{ ns}.$$

The logic jitter, a result of variations in the switching times of logic elements used to determine error, is typically 1% of the total logic delay. For a 100 ns logic delay, the logic jitter e_j is 1 ns.

Using these values, we have

$$T_g = 26.5 \text{ n sec}$$

Note that this guard time does not account for the ringing effect of satellite receiver filter which becomes important at high transmission rates. It is assumed that a 50 ns is needed for this effect pending on further survey of the state-of-the-art. To assure adequate time to prevent burst overlapping, it appears that a minimum guard time of 100 ns is required. Implementation of a 100 ns guard time system presupposes that ranging can be obtained very accurately so that the timing uncertainty due to ranging is very small. For example, the propagation path length difference between a zero elevation location terminal (41679 Km to synchronous satellite) and a 90° elevation location terminal (35786 Km) is 5893 Km. The difference in propagation time delay is about 0.02 seconds. If the receiver's ability to estimate the time delay is within $\pm$ 1%, the resulting time uncertainty is 0.2 ms. This very large (as compared to the 100 ns) timing uncertainty must be included in the guard time between packet burst unless the exact location of satellite and earth terminals are known through means at the expense of increased earth terminal complexity.

5.4.5 Low Duty Cycle, High Peak Power Technology

Preliminary survey of vendors of high power amplifiers indicates that technology applicable to the RMA traffic requirement of this study

is not a primary constraining factor to system implementation. Pulsed TWTA can be used for average power on the order of 100 watts or less. For higher power rating, a klystron can be used. To this end, it is of interest to note that although a spread-spectrum RMA system offers no system gain as indicated previously. It does offer the advantage of reducing the peak power requirement in both the transmitting terminals and the satellite. SS-RMA also offers additional advantages of AJ protection and spoofing. A trade-off between the cost of complexity of SS-RMA and the cost of using higher peak powered transmitter and transponder should serve to clarify the issue greatly.

5.4.6 TDMA Implementation of DVRA/DVDA

We now consider TDMA implementation of the hybrid destination-variable circuit demand access/packet random access technique proposed in this study. There appears to be no new implementation risk associated with this technique.

The TDMA system consists of dividing the time frame into smaller, nonoverlapping time slots. In any time slot, only one carrier accesses the satellite. In TDMA, usually one station acts as timing reference and sends periodic bursts without closed-loop control. The other stations in the network use closed-loop synchronization through the satellite to keep their burst transmissions within their time slots (a guard time is provided between time slots to cushion timing uncertainties due to satellite motions, slant range differences, etc.). The burst lengths are not necessarily the same since different traffic loads may occur at different stations. Reconfiguration of burst lengths at each station to accommodate traffic variations can be accomplished automatically. A brief description of a typical TDMA burst is in order: As shown in Figure 5.4.3 the guard time (on the order of 200 nsec.) is provided to ensure that successive bursts will not overlap. This is followed by a carrier-recovery/bit-timing recovery (CR/BTR) bit sequence. The number

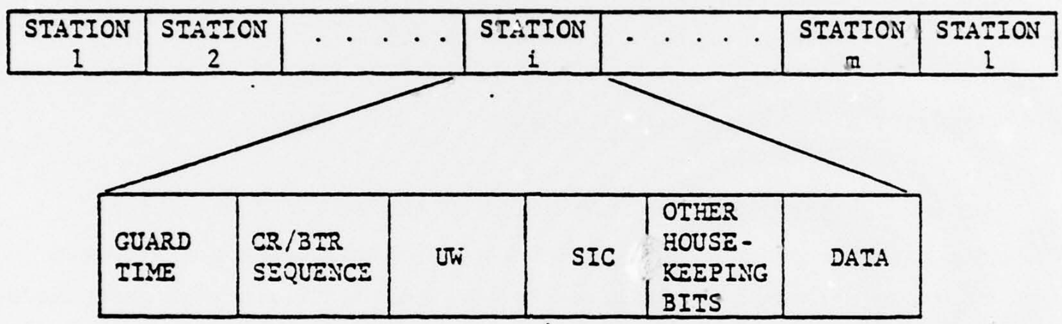


FIGURE 5.4.3 TDMA BURST DIAGRAM

of bits required for this sequence is the time required for the PLL to acquire this sequence. Since the PLL acquisition time is approximately the inverse of PLL bandwidth, which is about 1% of the bit rate at the present state-of-the-art, the bit sequence is about 100 bits for a 100-Mbps system. After a CR/BTR sequence, a unique word (UW) of about 20 bits is provided to enable the receiver to establish an accurate time reference in the received burst for the subsequent location determination of each data bit. Following UW, a word of 6 bits or so is transmitted as the station identification code (SIC) (UW can also be used for this purpose). It is important to detect the UW correctly; otherwise, the entire data burst will be lost. Other housekeeping functions such as orderwires, transmission bit-error rate sequence, etc., can be provided between SIC and data bits. The total preamble (all overhead bits) is usually 100-200 bits.

The inclusion of guard time and preamble bits along with the data bits in the TDMA frame means opertion at less than 100% bit rate efficiency. If C is the information bit rate, the actual burst rate R_b is given by

$$R_b = \frac{NP + CT}{T \quad NG},$$

where

P = number of preamble bits
T = frame time
N = number of slots (earth terminals)
G = guard time

The efficiency $e = C/R_b$ is then given by

$$e = \frac{CT - NCG}{NP + CT}$$

Figure 5.4.4 plots efficiency against number of earth terminals assuming:

$C = 200$ Mbps

$G = 200$ ns

$P = 200$ bits

While a long frame period is most efficient, the frame period introduces framing delay in addition to congestion queueing delay at the terminal. As can be seen in Figure 5.4.4 a 1 ms frame time provides high efficiency while introducing negligible delay for digitized voice purposes.

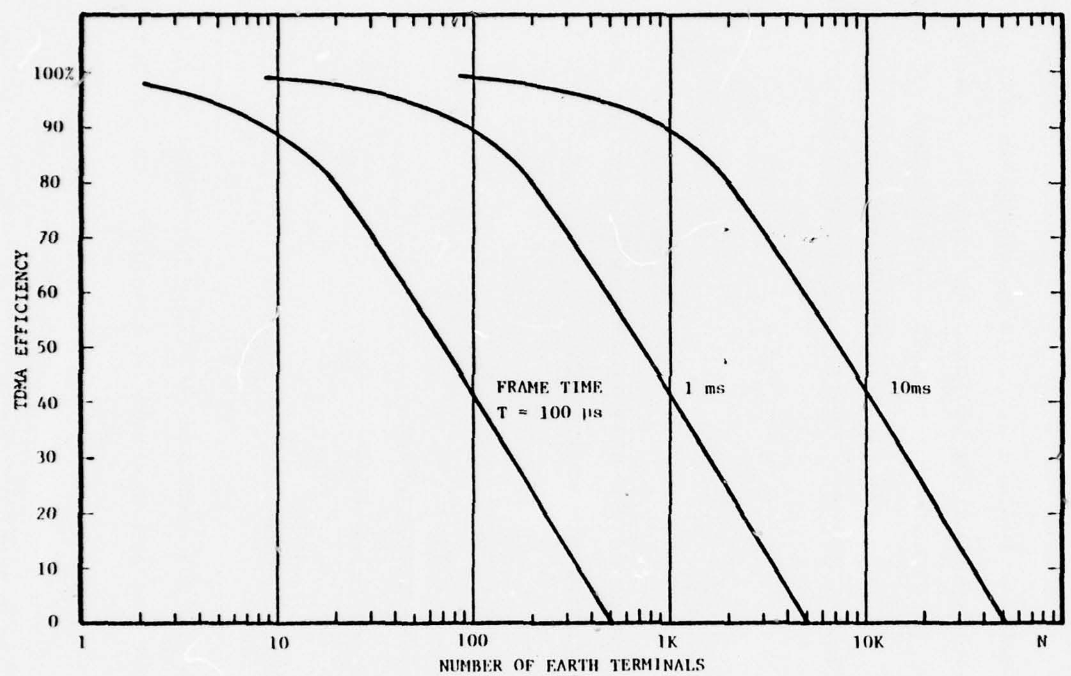


FIGURE 5.4.4 TDMA EFFICIENCY AS A FUNCTION OF NUMBER OF EARTH TERMINALS

5.5 COST OF EARTH TERMINALS AND SPACE SEGMENT

5.5.1 Cost Analysis Methodology

To facilitate the subsequent global analyses of cost and saving of replacing terrestrial communication networks with satellite communication systems, using DAMA techniques where the deployment of a large number of earth terminals is made to replace the access network as well as the backbone trunking, a cost model is proposed as illustrated in Figure 5.5.1. This section is concerned only with the assessment of curve A (annual earth terminal cost) and curve B (annual satellite cost), and the local optimization of the two. Detailed cost analyses are presented in Appendix B.

5.5.2 Earth Terminal Cost

The major components of the earth terminal in terms of the present technological approach are the following:

- Voice processing unit/modem
- Up/down converters
- High-power amplifier (HPA)
- Low-noise amplifier (LNA)
- Antenna
- Access control computer

of which the access control computer is highly dependent of the demand-access technique involved and will not be considered at the present time.

To allow subsequent intra-terminal as well as ET and satellite cost tradeoff analyses, component-cost characteristics must be established. Table 5.5.1 shows the respective characteristics to be collected for each component, with its potential suppliers as data base. [48]

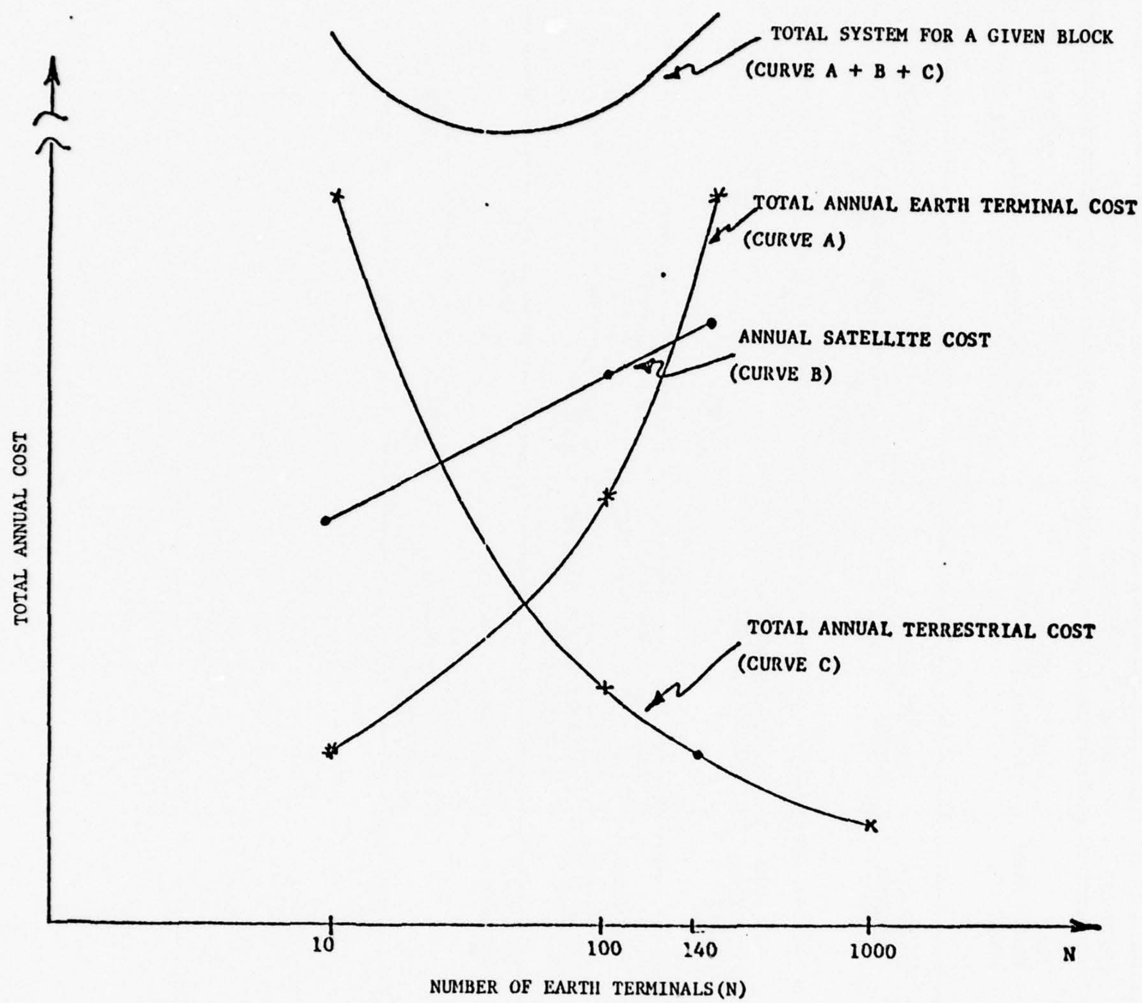
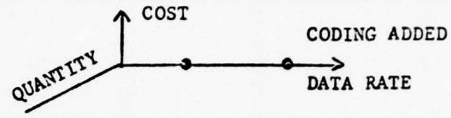
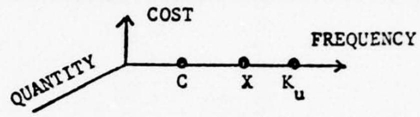
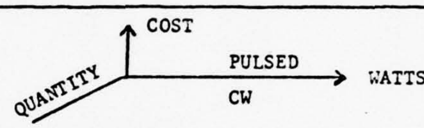
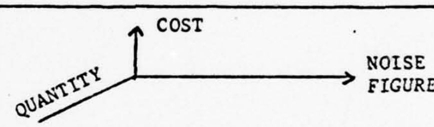
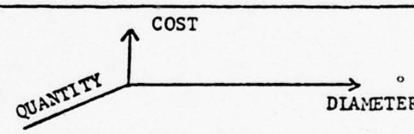


FIGURE 5.5.1 TOTAL ANNUAL SYSTEM COST AS A FUNCTION OF EARTH TERMINALS FOR A CANDIDATE SYSTEM

TABLE 5.5.1 COMPONENT-COST CHARACTERISTICS

COMPONENT	POTENTIAL SUPPLIERS	COST VARIABLES
1. VOICE PROCESSING/ MODEM	DCC, PHILIPS, GE, FUJITSU, GTE-ITALY	
2. UP-DOWN CONVERTER	COMTECH, MITEQ SCIENTIFIC-ATLANA	
3. HIGH-POWER AMPLIFIER (XMIT)	VARIAN, HUGHES ENERGY SYSTEMS	
4. LOW-NOISE AMPLIFIER (RECEIVER)	i) SOLID STATE: AVANTEK SCI(SCIENTIFIC COMM.INC) AMPLICAN, NARDA ii) PARAMETRIC: AIL,LNR COMTECK, SCI	
5. ANTENNA	i) LARGE: WDL ii) SMALL: WDL, ANDREWS, PRODOLINE, SCIENTIFIC-ATLANTA	

To assess the impact of ET sensitivity (G/T) on cost, ET's G/T is plotted as function of single-item cost of LNA and antenna for C, X and Ku-band in Figures 5.5.2, 5.5.3, and 5.5.4, respectively. In each of the figures the lower envelope constitutes a minimum cost curve. The figures also suggest that G/T is highly sensitive in cost for military grade X-band and is much less so for commercial grade C-band. In either case, the cost increases monotonically with the increase of G/T.

5.4.3 Earth Terminal Cost Optimization

The cost information presented in Figures 5.5.2 through 5.5.4 can be combined with the cost of high power amplifiers (HPA) to derive cost performance curves as a function of receiving terminal sensitivity G/T and terminal transmitting capacity (number of channels). The resulting curves are presented in Appendix B where single item costs of earth terminal (antenna + LNA + HPA) are plotted as a function of G/T for terminals capable of transmitting 1, 10, 100 and 1000 16-kbps channels. Examination of these figures reveals that minimum cost points are such that higher G/T (25 dB and up) terminals should be used for large number of channels in general, and lower G/T (20 dB and less) are optimal for smaller number of channels. In the derivation of these figures, assumption is made that equal cost prevails for pulsed HPA whose average power is the same as that of CW HPA. The assumption is due to the inclusiveness of vendor survey regarding to this matter, although pulse HPA are generally less costly than CW HPA at high power level (e.g., 1000 watts).

In addition to the cost of antenna, LNA and HPA, common equipment cost per earth terminal (excluding voice processing) are assessed at \$100,000, \$45,000 and \$50,000 for SCPC, FDMA and TDMA, respectively as shown in Table 5.5.2. Note that these cost figures are rough estimates based on literature study, not on vendor survey.

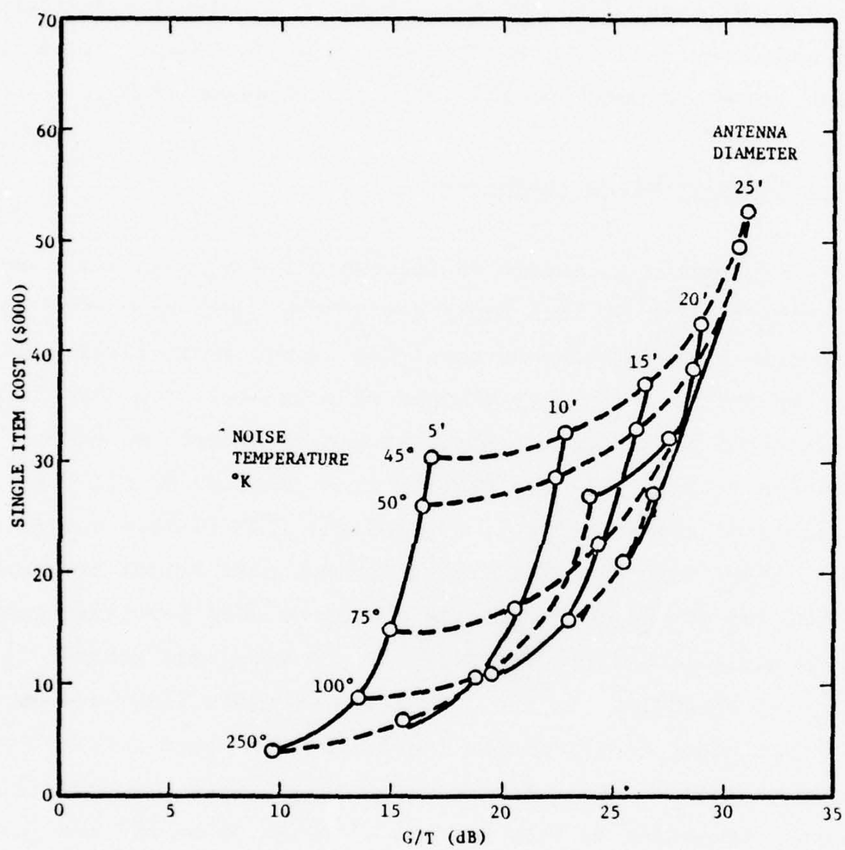


FIGURE 5.5.2 C-BAND ET (LNA + ANTENNA) COST

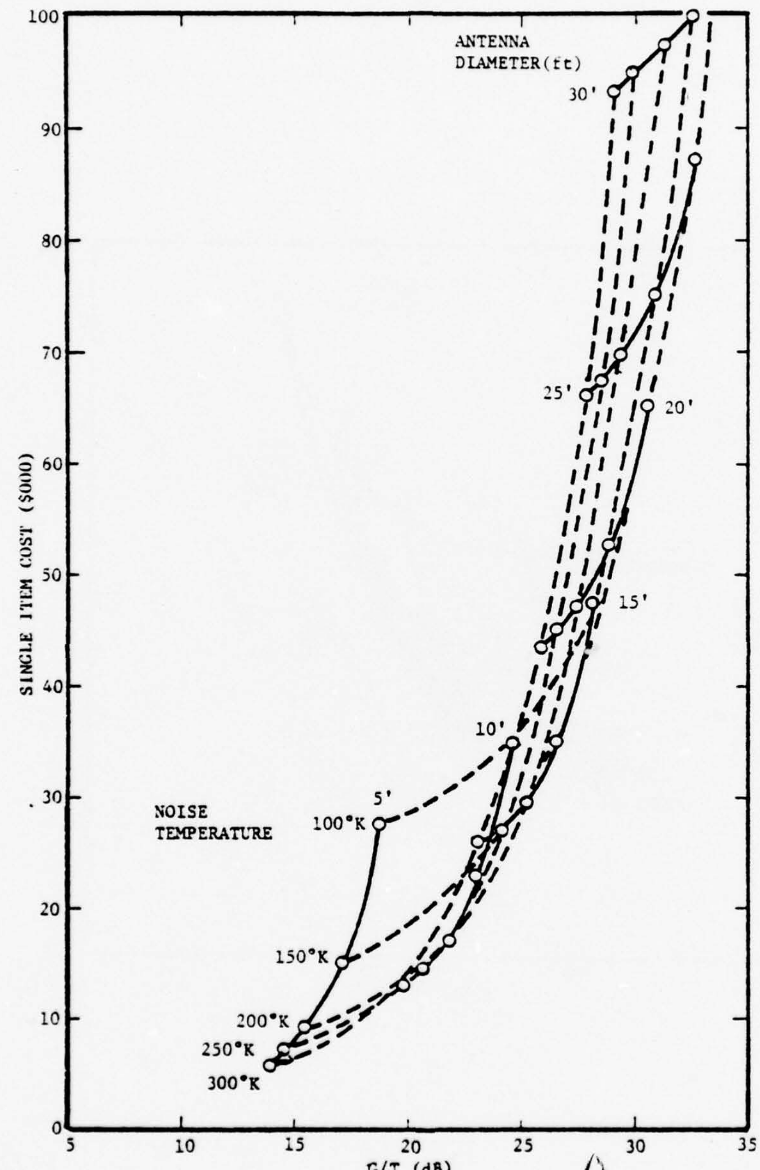


FIGURE 5.5.3 X-BAND ET (LNA + ANTENNA) COST

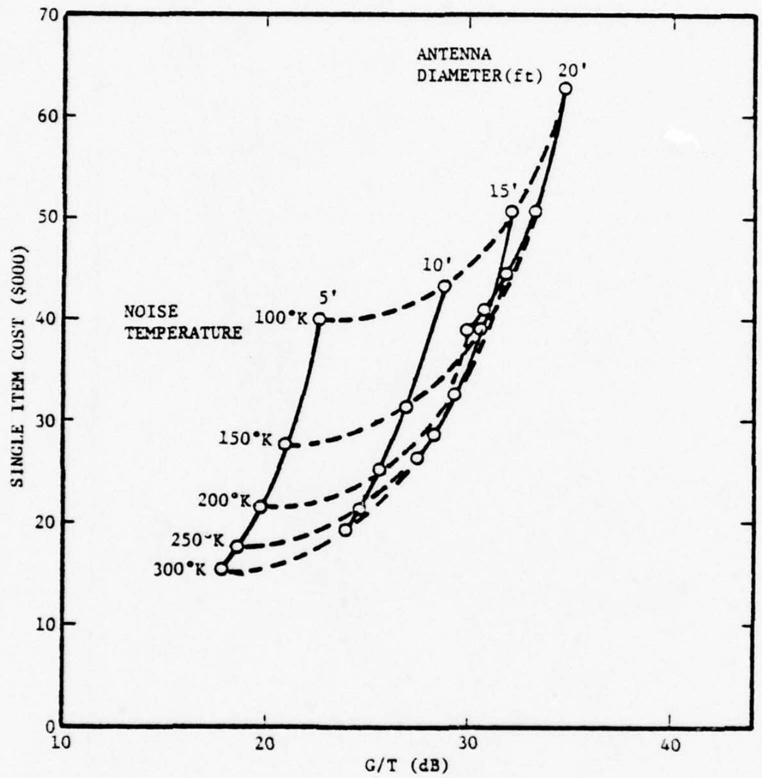


FIGURE 5.5.4 K_u-BAND ET (LNA + ANTENNA) COST

5.5.4 Space Segment Cost

The space segment cost can be estimated on a buy-or-lease basis.

Assuming that the satellite technology employed is fairly standard so that only the recurring costs are involved, the cost of production and emplacement of a geostationary satellite is given approximately by [25]

$$C_s(\$M) = 0.026\ (W_p)^{2/3}\ (1 + K + \frac{22238}{8000})$$

where:

W_p is the payload weight in pounds, and K is a constant determined by the payload sophistication. For example, the INTELSAT-V has a payload W_p = 4112.5 lb. Assuming K = 4, the single-satellite cost then is

$$C_s = 0.026(4112.5)^{2/3}\ (1 + 4 + 2.78) = 51.92\ M$$

Assuming a 10-year lifetime, the annual cost is then

$$C_{sa} = \frac{52.92}{6.45} = 8.21\ M$$

There is a total of 2137 MHz of bandwidth available in an INTELSAT-V, so that the annual cost per 1 MHz of bandwidth is

$$C'_{sa} = \frac{8.21\ M}{2137} = 38.4\ \$K/MHz$$

or the equivalent of \$1.38M per 36-MHz transponder. On the other hand, INTELSAT generally charges \$1M/36-MHz transponder for a spare (preemptible) to \$3M/36-MHz transponder (non-preemptible) in bulk leases. Table 5.5.3, using COMSAT data, shows a doubling of leasing charge for each 3 dB increase in EIRP.

TABLE 5.5.2 COMMON EQUIPMENT COST ESTIMATE

ITEM	COST		
	SCPC	FDMA	TDMA
Orderwire Source		3,500	3,500
Data Source	3,500	3,500	
Converter	3,000	3,000	3,000
Modem	20,000	20,000	25,000
Frequency Standard		2,500	2,500
Equipment Rack & Wiring	10,000	10,000	10,000
Orderwire Processing		1,000	1,500
Other Items	1,500	1,500	3,500
SCPC Equipment	62,000		
TOTAL	$100,000	$45,000	$50,000

TABLE 5.5.3 ESTIMATED ANNUAL CHARGES VS EIRP

NO. OF TRANSPONDERS	TRANSPONDER EIRP	ANNUAL CHARGES/TRANSPONDER
12	33 dBW	$1 Million/year
6	36 dBW	$2 Million/year
3	39 dBW	$4 Million/year

5.5.5 Satellite/Earth Terminal Cost Estimate

Using cost information obtained in this study, as discussed in Appendix B, annual system cost estimates on satellite/earth terminals are made with respect to the following multiple access techniques for C, X and K_u band:

SCPC
FDMA
TDMA
ALOHA
SLOHA (Slotted ALOHA)

For each of the three frequency bands, a composite minimum cost curve is constructed as a function of the number of earth terminals as shown in Figure 5.5.5. As an initial estimate in this task, it is assumed in the calculations that the annual satellite charge is proportional to its required bandwidth, which is assumed to be $1M/36 MHz for C, $1M/125 MHz for X, and $1M/240 MHz for K_u band, respectively. Under these conditions, the result of Figure 5.5.5 indicates a relatively moderate increase of annual system cost as the number of earth terminals N increases to about 100, but the cost escalates rapidly as N increases beyond 100. At N=100, the annual satellite/ET system charge for C, X and K_u band are $5.4M, $7.9M and $2M respectively. The relatively low cost of K_u band is probably due to the lower satellite transponder lease charge ($1M/240 MHz) on a per MHz basis assumed. However, the overall annual system cost in each case is dominated by the 20 man level O&M cost.

By excluding the O&M cost, a plot of annual minimum system costs is presented in Figure 5.5.6. Based on the cost model assumed, it is found that TDMA systems are minimum cost systems for the number of earth terminals N is less than 100. For N greater than 100, SCPC systems are minimum cost systems among the five types of access techniques considered.

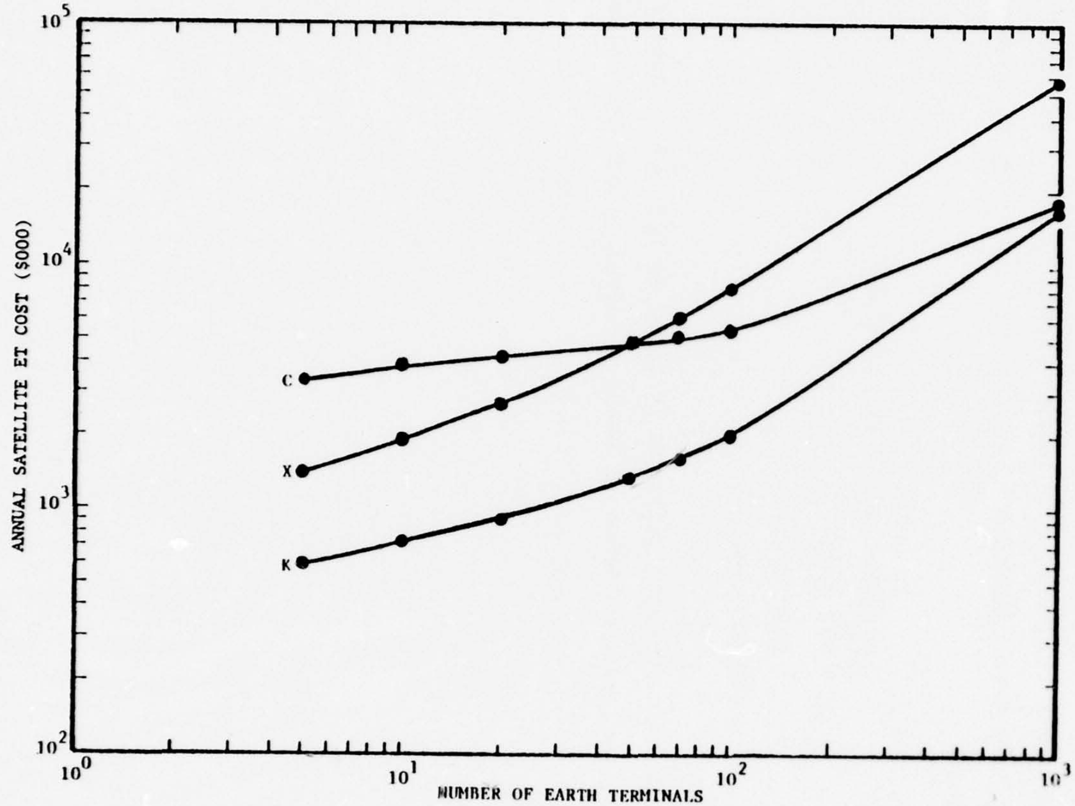


FIGURE 5.5.5 ANNUAL SATELLITE/EARTH TERMINAL SYSTEM COST EXCLUDING O&M COSTS

Since activation cost could vary more drastically than the true equipment cost in future systems, a plot of "equipment-only" annual system cost (i.e., not including system activation cost and O&M cost) is presented in Figure 5.5.7

In addition to annual satellite/ET system cost with O&M cost at 20 man-level per earth terminal. The annual satellite earth terminal system costs have been evaluated at various man level for C, X and K_u-band systems:

Case 1	-	0 man-year/ET	(Unattended)
Case 2	-	4 man-year/ET	
Case 3	-	8 man-year/ET	
Case 4	-	12 man-year/ET	

The labor cost in every case run is based on DCA cost document. The resulting cost versus number of earth terminals plots are documented in Appendix B.

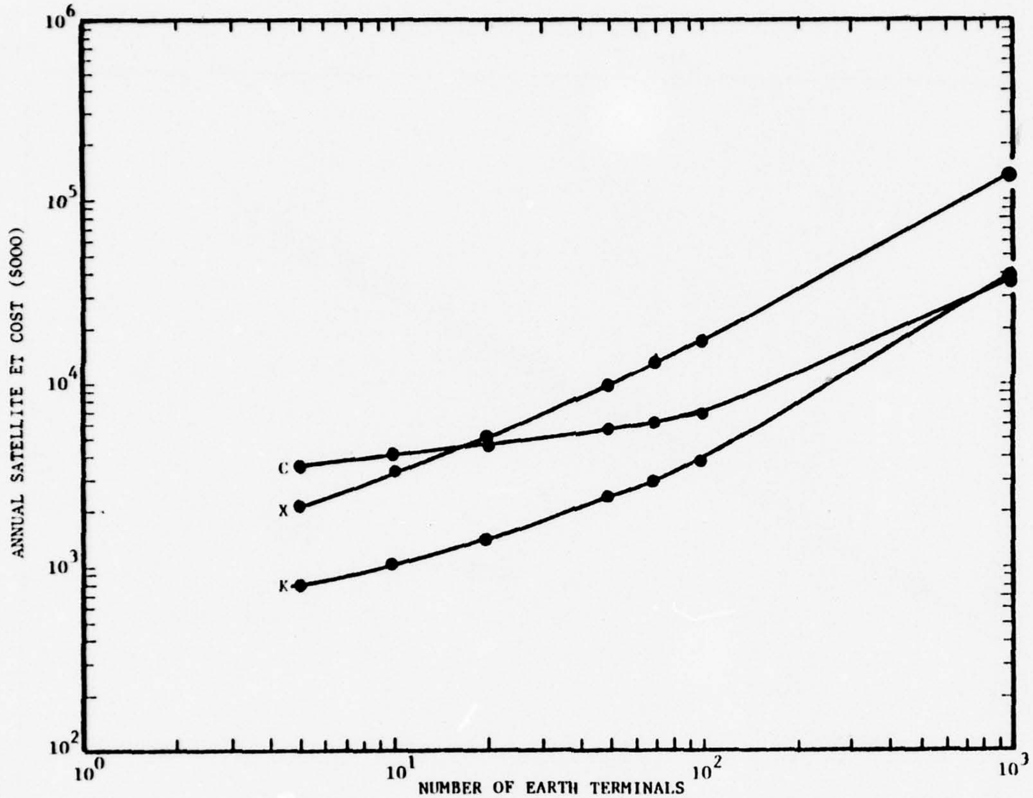


FIGURE 5.5.6 ANNUAL SAT/ET MINIMUM SYSTEM COST ESTIMATE

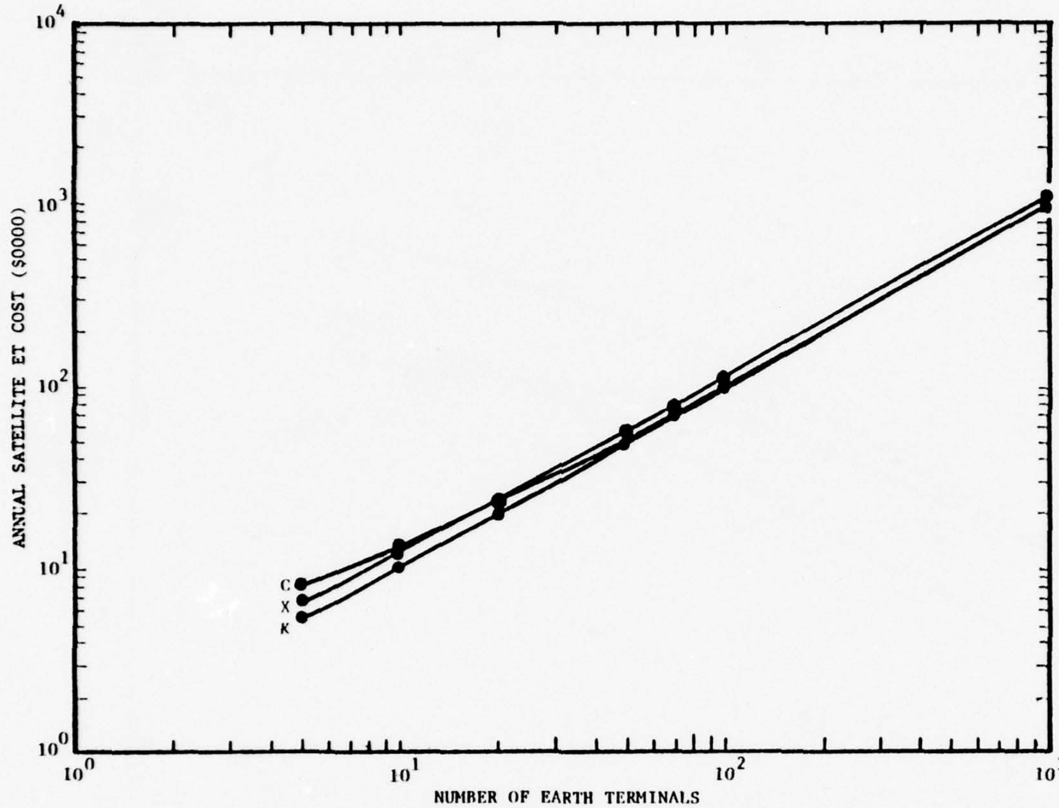


FIGURE 5.5.7 ANNUAL SAT/ET MINIMUM SYSTEM COST ESTIMATE (EQUIPMENT ONLY)

6 SYSTEM-WIDE COST TRADEOFFS

This Chapter presents the results of system-wide tradeoffs to determine the cost-optimal deployment of candidate DAMA techniques as partial or full replacement of the terrestrial transmission network.

This analysis is dependent upon selection of a large number of system and user requirement parameters. In order to reduce the parametric dimensionality problem to a manageable level to make the basic comparisons and tradeoffs, these parameters were grouped as follows:

Primary Design Parameters

- N = number of earth terminals
- A = MA/DA technique
- S = satellite/terrestrial traffic load split

Secondary Design Parameters (partial list)

- V = vocoder (voice digitization) type
- b = voice digitization rate (Kbps)
- t = packet time (sec)
- H = header length (bits)
- R = coding (rate and type)
- P_e = bit error rate ($0 < P_e < 1$)

Performance Parameters (partial list)

- P_L = packet loss rate
- D = packet delay (sec)
- W = setup waiting time (sec)
- Q = voice quality
- B = blocking rate

Implicit Parameter

C = satellite raw data capacity (Mbps)

Realization Parameters (partial list)

G/T = earth terminal figure-of-merit (dB/°K)

EIRP = satellite effective isotropicly radiated power (dBw)

M = manning level of earth terminals

The methodology behind this grouping is as follows:

1. Establish nominal requirements for all performance parameters.
2. Fix all secondary design parameters which directly satisfy the nominal performance parameters.
3. All remaining secondary design parameters are set to a nominal value based on present knowledge and judgment.
4. For the traffic model and for each candidate access technique (A), vary the number of earth terminals deployed (N) and the fraction of traffic (S) potentially carried by the DAMA satellite communication system.

Other ground rules chosen to simplify the scope of this baseline study are as follows:

- Single earth coverage satellite
- No onboard processing
- Consider further variations of assumed nominal parameters in this study as time and cost permit
- Use the predominant voice-only traffic

The goal of this procedure is to define the cost-optimum deployment region of earth stations (N) and traffic mix (S) for a broad range of DAMA techniques, and to approximately quantify cost/savings/complexity/risk factors for such deployment.

Figure 6.1 presents a flow diagram of this analysis. The total raw data capacity (C) required of the satellite is treated as an implicit variable in the following sense: Analysis of an access technique produces an explicit functional relationship between capacity and performance:

$$\text{Performance} = \text{function } [C, N, \text{traffic, vocoder, etc.}]$$

Then nominal performance (as applicable to the particular access technique) taken as a constraint implies a minimum capacity (C) which the satellite/earth terminal combination must achieve. This is the implicit (required) capacity. Determining C vs. N (for S = 1) for candidate access techniques is the fundamental step in efficiency/cost comparison of candidate DAMA techniques, since satellite EIRP and earth terminal G/T (which are the main contributors to system cost) depend mainly on C.

The boxes of Figure 6.1 correspond to relationships developed in other chapters of this report. The "Traffic Model" and "Incremental Terrestrial Communication Cost Savings" relationships are detailed in Chapter 3; the "Vocoder Tecnhique Performance" relationships are developed in Chapter 2; the "Access Technique Performance" relationships are developed in Chapter 4; and the "Space Segment... and Earth Terminal Realization Costs" relationships are developed in Chapter 5.

The terrestrial cost savings model (presented in Figure 3.3.1) is assumed to scale directly with split factor, S. That is, if only the fraction S of terrestrial traffic which could be carried by the satellite is used, then the cost saved is S x [Figure 3.3.1 savings].

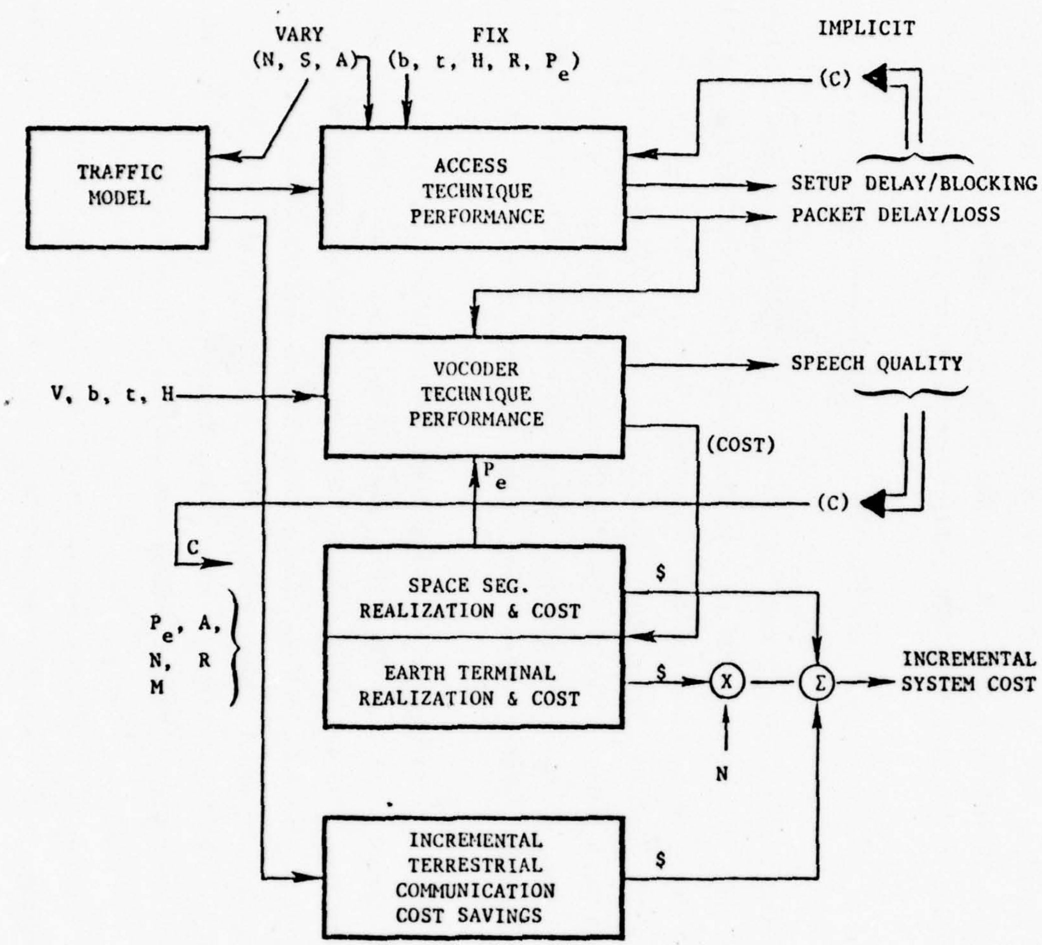


FIGURE 6.1 SYSTEM-WIDE COST TRADEOFF

As made clear in Chapter 4, only a few DAMA techniques are sufficiently efficient to be viable candidates. These candidates and the required capacity C (not counting overhead) for 16 KBPS voice digitization are

Access Technique	Capacity (MBPS)
DVRA	128
FVDA	244
DVDA	272

Fully Variable Demand Assignment was taken as a baseline technique for costing purposes. By subsequently varying the traffic split from 100% to 50% it was determined that the DAMA technique as reflected in C had very minor effect on the total space and earth segment costs.

As shown in Chapter 5 and Appendix B, the space and earth terminal segment costs are totally dominated by the manning level parameter. Since there is no obvious relationship between any other parameter such as access technique and manning level, the results are presented parametrically. It should be noted that the 20 man-year level is the current DCA practice. It reflects 24 hour manning by a crew of four, with one crew on training and one crew off duty.

The total annualized cost for transponder and earth terminal vs. number of deployed terminals and manning level and traffic level (100%, 50%) is presented in tabular form in Table 6.1. Also shown is the corresponding terrestrial cost savings.

The net annualized cost savings, determined by subtracting costs from terrestrial savings is presented graphically in Figures 6.2 and 6.3.

The strongest and most dramatic conclusion which can be drawn from these results is that

- Both the optimum deployment and overall cost savings are strongly dominated by earth terminal manning level.

TABLE 6.1 ANNUALIZED EARTH TERMINAL AND TRANSPONDER ACQUISITION AND OPERATION COSTS

N	Terrestrial Savings	0-ML	5-ML	10-ML	15-ML	20-ML
7	75	0.89	2.5	4.1	5.7	7.3
35	98	1.9	10	18	26	34
70	112	2.9	19	35	51	67
210	138	6.8	55	103	152	200

(a) 100% Traffic. (All figures in $\$ \times 10^6$)

N	Terrestrial Savings	0-ML	5-ML	10-ML	15-ML	20-ML
7	38	0.61	2.2	3.8	5.4	7.0
35	49	1.5	9.4	17	25	33
70	56	2.5	18	34	50	66
210	69	6.2	55	103	152	200

(b) 50% Traffic (All Figures in $\$ \times 10^6$)

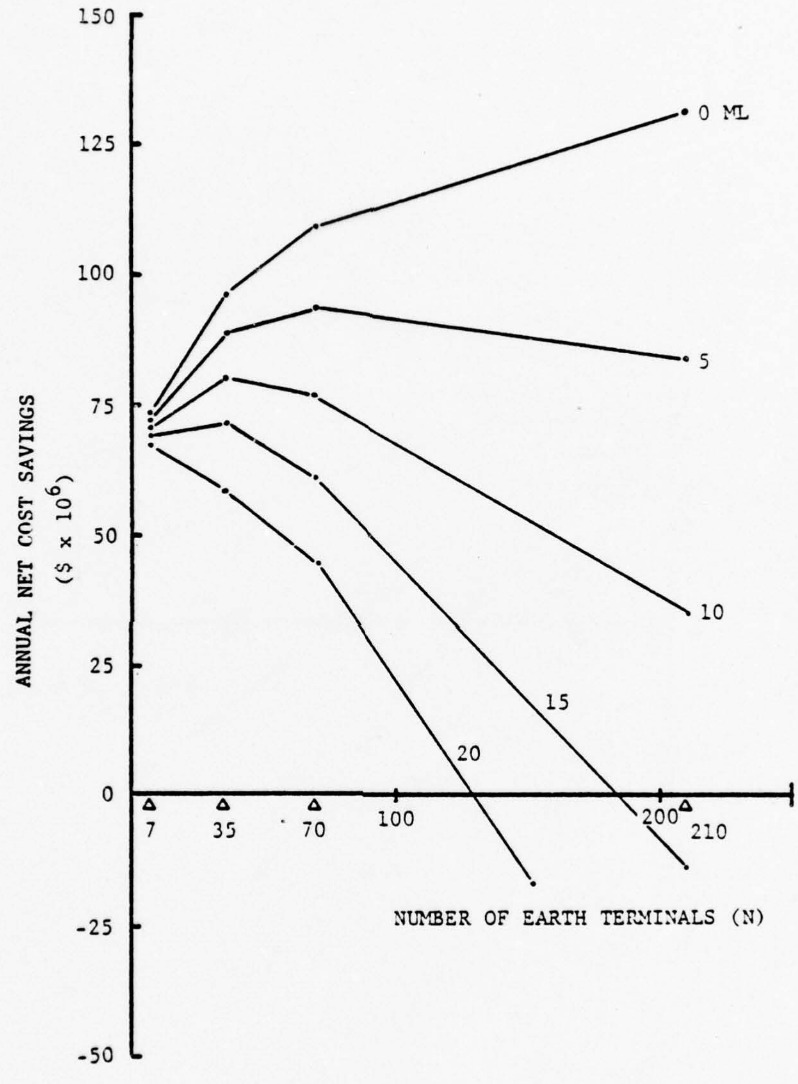


FIGURE 6.2 NET COST SAVINGS AS A FUNCTION OF NUMBER OF EARTH TERMINALS DEPLOYED AND MANNING LEVEL FOR 100% TRAFFIC SPLIT

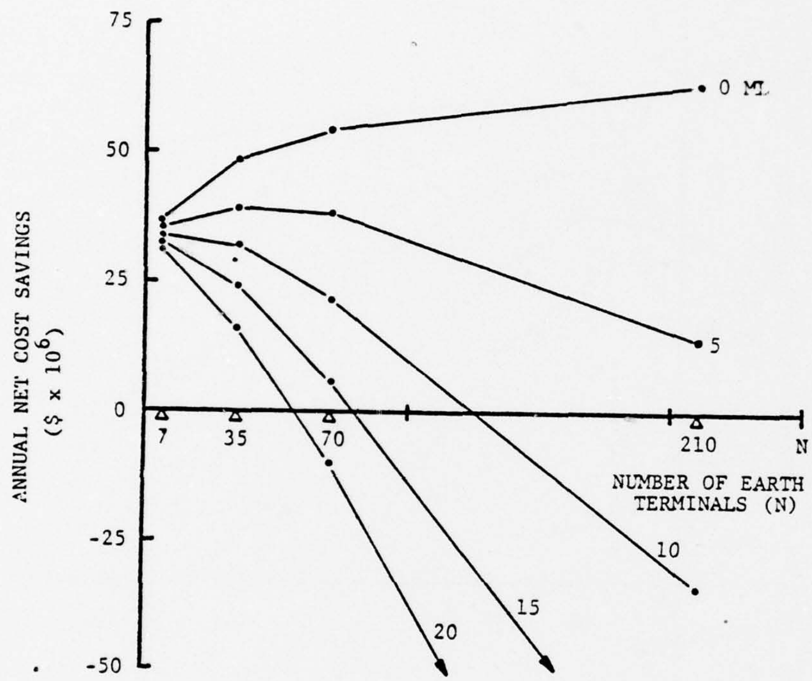


FIGURE 6.3 NET COST SAVINGS AS A FUNCTION OF NUMBER OF EARTH TERMINALS DEPLOYED AND MANNING LEVEL FOR 50% TRAFFIC SPLIT

At 100% traffic, and full (20 m) manning it is not cost-effective to deploy more than 7 earth terminals. As the manning level is reduced, the optimum deployment and savings both increase. At a manning level of 5 (which is probably a minimum allocation of one person/24 hrs. for a well-automated system) the optimum deployment moves out to 70 terminals- corresponding to replacing all terrestrial interswitch trunking.

If the traffic is 50% split (retaining 50% terrestrial fraction) the optimum deployment is only 7 terminals for all manning level of 5, the optimum deployment increases to 35 earth terminals (one per two switches).

Since the costs in Table 6.1 are not reduced significantly by 50% traffic reduction, we also conclude that these results are not sensitive to access technique changes which require capacity changes on the order of 2:1.

7 EVOLUTIONARY ISSUES AND APPROACH

7.1 INTRODUCTION

This study has been framed mostly in terms of the projected future traffic level and mix for DCS common-users the 1980-1990 time frame. There has been little emphasis on interfacing with the present DCS environment or bridging the gap "from now to then." The present chapter will emphasize these transitional problems.

We forsee the evolution of the traffic mix in the DCS as moving from a mixture of analog and digital to all-digital in the near-term (next 5 years) - at least in the backbone networks. Simultaneously, the switched non-voice traffic is moving toward all packet-switched (with the full deployment of AUTODIN-II). The problems of acceptable and bit rate efficient digitization of voice have resulted in a slower pace. The deployment of AUTOSEVOCOM-II will move the DCS toward more all-digital but still circuit-switched voice. The last stage of this evolution in the DCS in the longer term (10 years and beyond) will be toward all-packetized voice. The benefits of the bit rate savings, on the order of 2:1 for packetizing voice at the source (before encryption), push toward acheiving this final stage of evolution of DCS traffic.

The next section presents a proposal for a hybrid DAMA technique which will be adaptable to the evolution from mostly bit stream traffic requiring circuit switching, to a mostly packetized voice and data traffic. The following section outlines how the deployment of such a large scale DAMA system can be best deployed in light of the system-wide cost estimates of Chapter 6. Finally, the last section of this Chapter outlines how a basic earth terminal design, sized for lower traffic and bandwidth, can be converted to higher capacity and shorter wavelength bands.

7.2 HYBRID APPROACH

7.2.1 Background

Hybrid techniques are defined as those which combine elements of both circuit-switching and packet-switching. We have been motivated to propose and analyze hybrid techniques from several different aspects.

- No RMA technique has proved as efficient as circuit-switched techniques (i.e., directionally and fully variable DA) - except for CAPTURE which requires a special satellite, and DVRA (Section 3.3.8) which is a new innovation not previously considered.
- Isolated data packet traffic is not compatible with voice call oriented circuit-switching.
- Even if desirable for an integrated data/voice network, digitized voice may not occur universally in packetized form.
- There may be continuing requirements in the DCS for nonvoice bit stream data (e.g., digital FAX, TV etc.) as well as end-to-end bit stream COMSEC secure voice.
- In addition to the common-user network the DCS contains many special interest (private line) circuits. A hybrid approach which can also accomodate these circuits would lead to a fully integrated DCS.

Clearly a hybrid technique is needed. In Section 4.4.8 the directionally variable random access (DVRA) method was shown to be the most efficient RMA technique investigated in this study. In Section 4.3 directionally variable demand assignment (DVDA) and fully variable demand assignment (FVDA) were found to be the most efficient circuit-switched DAMA techniques, with FVDA becoming superior only for more than

about 200 earth terminals. For these reasons along with further control and security advantages to be discussed in Chapter 8, a hybrid DVDA/DVRA technique is proposed as the preferred DAMA approach for the future DCS. The following section outlines the combined or hybrid protocol, and describes how it could adapt to follow evolution from primarily a circuit-switcher of bit steams to primarily a packet-switch.

7.2.2 Adaptive Hybrid Approach

For N earth terminals, the total satellite transponder capabity C is split into N parts C_i (not necessarily equal):

$$C = \sum_{i=1}^{N} C_i .$$

For definiteness we assume that this split is effected by a TDMA frame of T sec structured with subframe time slots T_i proportioned as:

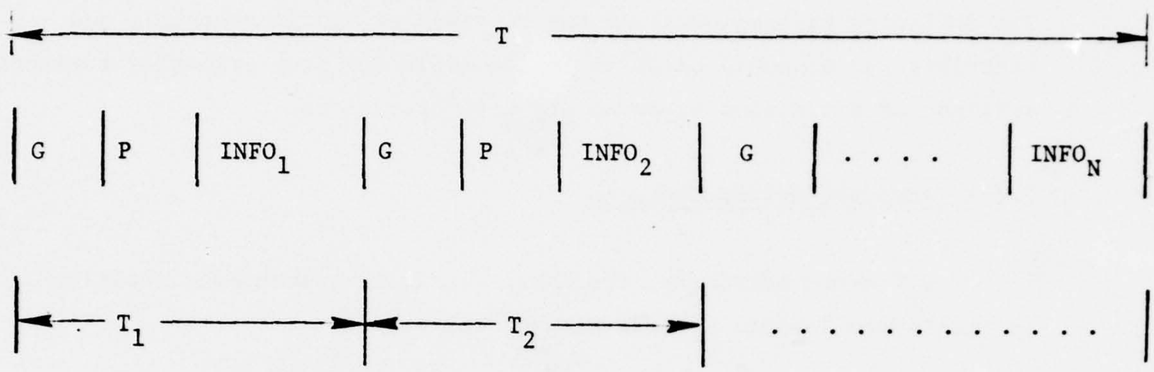
$$\frac{T_i}{T} = \frac{C_i}{C} .$$

The overall frame structure is illustrated in Figure 7.1. Further details of preamble bits and guard times are discussed in Chapter 6.

Each ET generates a carrier burst of preamble bits and information content bits in its alloted time. The adaptive hybrid DVDA/DVRA protocol is a further structure within the block of "content" bits. The following subsections consider content-switched DVDA, pure packet-switched DVRA and, finally, an adaptive combination, respectively. For discussion purposes we shall assume that the capacity split among earth terminals is equal, i.e.,

$$C_i = C/N$$

(but this is not necessary).



G = Guard time

P = Preamble bits

FIGURE 7.1 TOTAL FRAME STRUCTURE

7.2.2.1 Pure DVDA

The information content block of bits which a given ET bursts every frame is of size:

$$I_i = TC_i = TC/N \text{ bits.}$$

This block is further divided into L groups or words. Each of these words represents an outgoing channel or half-duplex circuit with the "other end" freely assignable to any destination (This particular version of DVDA is called destination variable). Thus, from the point of view of a given earth terminal, there are L outgoing bit stream channels, each operating at a standard speech digitization rate b bits/sec:

$$bL = C/N$$

A feasible protocol for call assignment of these L channels is as follows:

a. An "idle channel" bit pattern is put on all idle channels

b. To set-up a call to a given destination, an idle channel is selected; the idle pattern removed; the address of the destination sent over the channel as a set-up request.

c. The destination ET upon receiving and recognizing the request selects an idle return channel and addresses the calling ET; also supplying the channel number to identify the particular call. This completes the set-up.

d. Other digital messages for hangup, answer, ringing, busy, preemption, etc. can be defined.

7.2.2.2 Pure DVRA

For pure packet DVRA, the information content block of bits transmitted by a given ET is not further structured. It simply forms a TDMA burst channel of average throughput rate C/N bits/sec.

Since every switched call begins with set-up information (i.e., called telephone number), a set-up procedure can be established whereby a set of inter-ET call set-up message packets are defined. In fact, the standard CCIS message set can be used with the addition of the following information:

a. Explicit identification of the destination ET and source ET.

b. A call index system whereby each new call is given an index or ID which distinguishes it from all other calls in progress.

The initial "handshake" or exchange of CCIS message packets between ETs establishes an ID for a call. All subsequent digitized voice packets for that call need contain only that ID as a header.

By examining the headers of arriving packets on all N-1 other burst channels, an ET can distinguish CCIS packets from voice packets and (a) act on CCIS messages addressed to it, and (b) accept and route voice packets addressed to it onto corresponding access trunks.

7.2.2.3 Destination-Variable DA/RA Hybrid

Combining the destination-variable circuit-switched DA and packet-switched RA concepts of the previous two subsections can be effected by an approach similar to the SENET concept. As illustrated in Figure 7.2, the time slot for a single ET burst is divided into guard time, preamble bits, and information content bits. The I information content bits are further subdivided into two blocks with the X bits forming circuit-switched channels and Y bits forming the packet broadcast channel for a given ET, where:

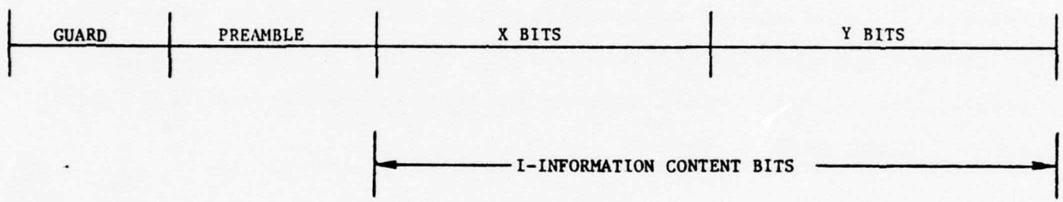


FIGURE 7.2 SINGLE SLOT HYBRID FRAME STRUCTURE

$$I = X + Y$$

and

$$I = CT/N,$$

in the equal traffic case.

Rather than independently operate the DVDA and DVRA protocols (described in the previous subsections) in the X and Y bit blocks respectively, the set-up protocol could be streamlined by using CCIS-type packet messages in the Y-bit packet channel to set up both bit stream calls in the X-block stream channels and packetized voice calls in the Y-block packet channel.

The establishment of the boundary between the X and Y blocks provides the mechanism for evolution from an almost all bit-stream operation (e.g., $X/Y > .9$) to an almost all packet operation (e.g., $X/Y < .1$). Several options are available for the degree of demand or timeliness involved in changing the X-Y boundary:

a. Nonreal-time in response to shift of historical demands; or

b. Real-time adaptive in response to current traffic demands;

c. Per-call expansion/contraction of X to set-up/remove bit stream channels in response to actual call set-up requests.

Either of the last two alternatives could make the evolution of operation from mostly bit-stream to mostly packetized data/voice fully automatic.

7.3 DEPLOYMENT EVOLUTION

From the system-wide deployment cost optimization in Chapter 6, it is clear that unless the earth terminal operation can be nearly 100% automated it is not cost-effective to deploy more than 7 terminals for replacement of all interregional trunking in the CONUS model.

If the destination variable packet/circuit hybrid technique is employed, it happens that a system sized for the full traffic load of 8000 erlangs with all-packets will carry about half as large a load when the load is all bit-streem (circuit-switched). Thus, a seven terminal system sized for a fully packetized ultimate load, can carry up to 1/2 that ultimate (erlang) load at an earlier time when the traffic is smaller. We have seen no model which forcasts the traffic level and packet/circuit mix growth paths for the future DCS, and thus can not determine if this characteristic if really a good growth match. The next section will discuss the feasibility of increasing the capacity and/or operating band of earth terminals once deployed.

After deployment of the first group of seven terminals, the actual O&M costs experience can be used as a guide to determine the effectiveness of further deployment. Thus, for example, if the O&M costs experience corresponds to the 10 man level from Figure 6.2 we would expect to gain an additional $9 million annual savings by deploying about 35 terminals.

On the other hand, the traffic potentially carriable by the satellite system could be split to allow only a fraction be carried by satellite - thus preserving a terrestrial backbone in case of satellite failure. Figure 6.3 shows that if this split is 50%, a 10 man-level O&M cost experience shows, it would <u>not</u> be cost-effective to extend the deployment beyond the initial seven earth terminals.

7.4 TECHNOLOGY EVOLUTION

Until recently, the assignment of frequency band for satellite communications was based on the user characteristics. For example, the low-data-cycle, military users with constraints on terminal size, weight and mobility are served at UHF, military users with trunking requirements are served at X-band; and commercial users are presently being served by C-band. This separation of user types by frequency band has probably occured because the historical development of UHF and SHF has coincided roughly with the development of user requirement. However, the demand for UHF-type service apparently will grow and soon will outstrip the ability of the limited UHF band to support such service. Futhermore, the characteristics of many of the X-Band users are changing so that netting of high peak bandwidth, low duty cycle users is required. Thus, the distinction of user type seems to be gradually blurring and it is important to design DAMA systems which are capable of handling a mix of users which would have been previously segregated. As the military use of satellites continues to grow, it is equally important to design systems with capability of accomodating capacity growth and technological upgrading, so that the life cycle cost of the system can be kept at a minimum. At issue are questions such as:

- Can C-Band and X-Band earth terminals be upgraded to provide X-Band K_u-Band service with any economical feasibility?
- Can small capacity earth terminals be expended to have a larger capacity without incurring huge cost?

These questions can not be answered in general terms and may have to be considered on a system by system basis. It suffices to say that systems such as SCPC, which have modular construction characteristics, render easier capacity expansion. As for changing the frequency band of operation, all components beyond IF stages will need some kind of modification or replacement because they are frequency optimized (feeds and RF amplifiers

are examples). Furthermore, the surface accuracy of an antenna dish designed for C-Band is usually not good enough for X or K_u-Band, causing the antenna efficiency to drop. A system by system cost-performance trade study is needed to resolve these issues. In the absence of such study here, it is conjectured that for small structure earth stations where mechanical cost is a small portion of the total, replacement of all frequency dependent components including antenna dish may be more feasible. While for earth stations with large mechanical structures such as HTs (Heavy Terminal), where mechanical cost is a large portion of the terminal, upgrading the existing components may be more feasible. These observations seem to apply to future designs as well, that is, for smaller terminals, design to cost (and performance) should be the rule. For large terminals, the extra cost is probably justified to upgrade such components such as an antenna dish to perform at highest planned future frequency of operation, for example, design to K_u-Band tolerances even though the current requirement call for only C or X Band.

8 CONTROL ISSUES

8.1 INTRODUCTION

Description and analysis of the RMA protocols and the Chapter 7 hybrid protocol proposal touched on a number of control structure issues. The purpose of the present chapter is to give a systematic summary and comparison of these control structure and related issues for these alternative protocols. The major control structure issues are in the following areas:

- Vulnerability to instability/lockup
- Central vs. Distributed Control
- Sensivitity to Traffic Shifts/Model Error
- Communication and Traffic Securability
- Vulnearbility to Jamming/Spoofing

The following sections will briefly define and discuss, in general, each of these problem areas, noting where appropriate protocols with particularily good or difficult characteristics in that area. Finally, a concluding section will provide a tabulated comparison of DAMA/RMA/Hybrid protocols studied against these issues.

8.2 INSTABILITY/LOCKUP

The original ALOHA concept and subsequent slotted ALOHA concept (4.4.1 and 4.4.2) both are based on the idea that packets lost by collision are to be retransmitted until successful, and both suffer from the problem of instability [13]. The instability occurs because the channel can become flooded with mutually destructive retransmission attempts. Attempts by several researchers to impose additional structure which estimates the channel traffic and "throttles" the inputs [45] have yet to prove completely successful. Implementing a special CAPTURE processing satellite (Section 4.4.3) not only improves efficiency but stabilizes the protocol.

The emphasis in this study has been on the voice control rather than the data context in which the original ALOHA protocol was invented. With synchronous satellite up-down propagation delay of about 280 miliseconds, infinite retransmission of voice packets is not at all appropriate. Thus, we were motivated to consider no-retransmission and finite-retransmission versions of ALOHA. For no-retransmissions or 1 or 2 retransmissions, as seems appropriate for voice, the instability problem completely disappears.

No other protocol studied displays instability or lockup.

8.3 CENTRALIZED VS. DISTRIBUTED CONTROL

For defense applications it is particularly important that the communication network connectivity not be vulnerable to failure or destruction of a single earth terminal or control node.

To discuss this issue, several different kinds of control can be distinguished:

a. Per-call or per-packet demand assignment
b. Timing
c. Traffic-shift reallocation of resources

The per-call or per-packet demand assignment is the most critical. Fortunately, virtually all protocols considered in this study have either naturally distributed control in this sense, or can be so designed.

Timing is by nature "centralized," i.e., established by some one source, yet it is feasible to provide all earth terminals with the capability of generating timing, and to establish an order of succession in the event of failure.

Traffic-shift is theoretically not needed for some protocols (e.g., ALOHA derivatives) and is needed for others (e.g., fixed and destination-variable assignment). Like power control changes in the DSCS-III/RTAC environment this level of control is not critical on a call-by-call basis, and furthermore is not associated with any particular earth terminal.

8.4 SENSITIVITY TO TRAFFIC SHIFTS

A theoretical advantage of the ALOHA-based RMA techniques and fully variable demand assignment over fixed-assignment and destination-variable demand assignment is an insensitivity to shifts in the distribution of traffic. All assignment techniques are vulnerable to increases of the aggregate traffic load beyond the design load. As discussed in Chapters 4 and 7 the DVDA and DVRA hybrid can be made responsive to traffic distribution shifts by use of a variable frame TDMA structure.

8.5 COMMUNICATION AND TRAFFIC SECURABILITY

Because of the broadcast nature of a satellite channel, any special COMSEC problems or properties of the protocols become important. The basic COMSEC issues are:

- Communication security (privacy)
- Traffic Security
- Authenticity

All of the protocols can in principle be secured in the privacy sense. If the information content bits of packets or bit stream calls are encrypted either end-to-end by the users or on a link basis between the earth terminals, this goal is achieved.

Traffic security, i.e., keeping the network traffic flows and levels secure, is particularly difficult with the packet-contention (ALOHA type) systems. While the content and even header/routing information can be encrypted, the level of network activity is plainly visible. Some activity could be hidden by placing fake "camouflage" traffic on the network - but at the expense of degrading performance. The destination-variable techniques including the DA/RA hybrid proposed in Chapter 4 is especially easy to secure for both traffic and communication because each ET "owns" a dedicated broadcast channel which can always be "on" with a running cipherstream.

Proper encryption of all information and control signals with time-changing methods also insures the authenticity of all messages received.

8.6 JAMMING/SPOOFING VULNEARABILITY

Another important area of the DAMA/RMA protocol control structure is its vulnerability to intentional or unintentional interferences, jamming, or spoofing. Of particular concern here is the issue of whether the control or assignment mechanism of a particular protocol is more vulnerable than the communication link itself.

For example, the orderwire channel used to allocate and deallocate channels in SCPC FVDA would be more vulnerable to jamming attack than the communication channel spectrum unless it were spread to at least as much bandwidth. Encryption, and/or the spread spectrum code on an orderwire channel can secure it against a spoofing attack.

The original (infinite retransmission) ALOHA protocols are especially vulnerable to jamming-induced instability. The jammer need inject only enough low duty cycle pulsed power to create a few bit errors per packet thereby triggering enough retransmissions to induce self-sustaining

saturation of the channel. The no-retransmission and finite-retransmission modifications introduced in this study (see Section 4.4.6) do not suffer from this kind of vulnerability.

The destination variable DA/RA hybrid proposed approach of Chapter 4 has no orderwire structure outside the per ET communication channels. The TDMA implementation was suggested chiefly because of its flexibility in reallocating capacity among ETs, but like any TDMA system is vulnerable to jamming attack on its timing. A less flexible mode of channelization which could be used in a stressed mode is CDMA. Alternately the timing could be made more robust by retaining bit timing for each slot coherently from frame-to-frame in memory along with AGC level. Detailed evaluation of cost and effectiveness of such design alternatives is beyond the scope of the present study, but should be considered in more depth.

8.7 PRIORITY/PREEMPTION OPERATION

The DCS Common-user network operation has historically been designed to accomodate a call priority structure whereby a call can preempt established calls of lower priority to gain access to fully occupied equipment, including the called number station equipment. The goal is to guarantee that the highest priority calls will always get through the network, even when the network is loaded with lower priority traffic. The issue to be addressed is to what extent are the alternative DAMA/RMA techniques compatible with priority operation.

First it can be noted that the pure circuit-switched DAMA techniques i.e., DVDA and FVDA are clearly fully compatible with priority/preemption operation; the unit of assignment is a circuit which can as easily be preempted and reassigned.

On the other hand the pure packet contention RMA protocols (pure and slotted ALOHA with and without retransmission) are not compatible

with traditional preemption since no resource is dedicated to any one call. By definition no calls are blocked by these protocols. Service (speech quality/intelligibility) degrades for all users with overload.

An approach to this problem is to institute a form of load control such as imposing a limit to the maximum total number of calls allowed on the system. To set up a call request arriving when the system is "full" would then require preemption of a lower priority call (beginning with the lowest).

Packet-oriented and message-oriented reservation systems are naturally adapted to a priority structure, but have not been emphasized in this study because of the incompatibility of the propagation delays involved with digitized voice requirements.

Finally, as discussed in Section 4.4.7, the proposed destination-variable random-access scheme is compatible with a priority structure by utilizing a priority queue system to insure nonloss of highest priority packets. The hybrid DVDA/DVRA technique proposed in Chapter 7 could be so structured for its packet traffic with conventional preemption used for its bit-stream traffic. If the boundary between bit-stream and packet traffic is continuously adapted to the call traffic mix at each ET, the boundary movement algorithm could impose maximum/minimum percentages on the mix to insure sufficiency of resources for both circuit and packet high priority traffic. Similar considerations apply to the traffic-adaptive movement of the relative sizes of time slots assigned to earth terminals. Specifically the capacity assigned to any one ET should not be allowed to become too small to handle a sudden burst of high priority traffic through that terminal.

8.8 SUMMARY OF CONTROL STRUCTURES

Table 8.1 provides a summary of overall control structure characterization/comparison of the voice RMA alternatives along with the non-RMA

DAMA alternatives and the proposed hybrid alternative of Chapter 7. Some of the comparisons are necessarily qualitative because of dependence on variable design choices and detailed analysis and simulation beyond the scope of this broad comparison.

TABLE 8.1 DAMA CONTROL STRUCTURE COMPARISON SUMMARY

PROTOCOL		Unit of Access	Stability	Control Distribution	Sensitivity to Traffic Shifts	Sensitivity to Control Jamming/Spoofing	Securability of Traffic [3]	Priority Control
RMA								
Pure ALOHA:		Packet		Distributed policy; no timing coordination	Low		Poor	Only by external load control (centralized)
Infinite Retransmission	(3.3.1)		Unstable			High		
No-Retransmission	(3.3.6.1)		Stable			Low		
n-Retransmission	(3.3.6.2)		Stable, n<7			Medium		
n-Copy (1-try)	(3.3.6.3)		Stable			Low		
Spread Spectrum	(3.3.6.4)		Stable, n<7			Low		
Slotted ALOHA:				Distributed policy; slot timing coordination required	Low			
Infinite Retransmission	(3.3.2)		Unstable			High		
No-Retransmission	(3.3.6.1)		Stable			Low		
n-Retransmission	(3.3.6.2)		Stable, n<7			Medium		
n-Copy (1-try)	(3.3.6.3)		Stable			Low		
Spread Spectrum	(3.3.6.4)		Stable, n<7			Low		
Capture ALOHA:				Distributed policy; can be implemented w/wo time slotting	Low			
Infinite Retransmission	(3.3.3)		Stable			Medium		
No-Retransmission	(3.3.6.1)		Stable			Low		
n-Retransmission	(3.3.6.2)		Stable			Medium		
n-Copy (1-Try)	(3.3.6.3)	Packet	Stable			Low		
Destination Variable RA	(3.3.8)	Packet	Stable	Distributed policy; timing coordination for TDMA implementation	Medium[1]	Low	Excellent	By priority Queues
NON-RMA DAMA								
Directionally Variable DA (Task I Report 4.3.2)		Circuit	Stable	Distributed policy; timing required	Medium[1]	Low		Conventional Preemption
Fully Variable DA (Task I Report 4.3.3)		Circuit		Central or distributed policy; timing for TDMA, power control for SCPC	Low	Variable[2]		
NON-DAMA MA								
Fixed Assignment MA (Task I Report 4.3.1)		Circuit		Timing for TDMA; power for SCPC	High	None		
HYBRID DAMA								
Hybrid DVDA/DVRA (4.2)		Clr/Packet	Stable	Distributed policy; timing for TDMA	Medium[1]	Low		Mix of priority and conventional preemption

NOTES: [1] DVRA and DVDA hybrid can be made insensitive to traffic shifts by reallocation of TDMA frame fraction.
[2] Jamming sensitivity of FVDA setup protocol depends on design choices. Can be made low.
[3] All techniques can be secured for communication privacy and traffic source/destination privacy. Only traffic level privacy is a function of DAMA technique.

9 IDENTIFICATION OF ADDITIONAL RESEARCH

9.1 DESTINATION VARIABLE RANDOM ACCESS BASED STUDY/EXPERIMENT

The destination variable random access (DVRA) technique for packetized voice originated in this study (4.4.8), is the most efficient and promising RMA technique investigated. The corresponding circuit or bit stream destination variable demand access (DVDA) is nearly the most efficient circuit-switched technique studied (4.3.2). We believe that DVRA and the DVRA/DA hybrid system proposed as the preferred evolutionary approach for the DCS (7.2) merit serious consideration and further study.

9.1.1 DVRA Terminal Design Problem Areas

The queuing analysis used in this study to determine packet loss rate versus utilization for a DVRA terminal was based on simplifying assumptions that:

- Did not distinguish between packets of separate calls,
- Assumed purely random input and output packet streams,
- Did not consider priority effects, and
- Did not analyze separate voice/data packet treatment.

Going beyond these simplifying assumptions, the analysis is linked with analysis and tradeoff of numerous system design alternatives. The proposed DVRA technique is at present more of an abstract concept than a system design. For example, an actual DVRA terminal design must include a packet buffer management algorithm, which simultaneously provides for:

- Sufficient randomization of packet losses within a call, and sufficient equalization of losses across calls,

- Handling of priority, and
- A mix of data and voice packets, with extremely low loss rate (overflow) for data packets, and a correct balance between delay and loss for voice packets.

The design and full analysis/simulation of the packet buffer management algorithm with the above goals also depends critically on a better understanding of digitized voice performance with packet losses and delay, and on packet generation statistics. For example, the sensitivity of communication efficiency and quality to a measure of randomization of packet losses should be quantified. Thus, a sound design of a DVRA terminal packet buffer management algorithm should be based on an integrated combination of digitized voice study and experimentation, and analysis and/or simulation of system design alternatives. Finally, any design, which is so different from previous approaches, requires a realistic field trial before it will be widely accepted.

9.1.2 Phased Design Study/Experiment

From the above considerations we proposed the following phased design development plan for the DVRA voice packet-switched approach.

Phase I: Basic Design Origination/Evaluation

This phase is oriented toward further development of the basic DVRA concept. The goals of this phase are:

(a) To select the best DVRA terminal buffer management algorithm and call setup procedures to provide for:

- Sufficient randomization,
- Prioritization,
- Data/Voice mix.

(b) To determine performance/design parameter tradeoffs.

These goals can be achieved by an approach that evaluates design alternatives by analysis and simulation based on packetize voice performance models developed specifically for this system environment.

Phase II: Design Refinement

The goals of this phase are:

(a) To refine the voice models and design parameters.

(b) To evaluate/demonstrate the feasibility and desirability of the system design short of employing actual satellite/earth terminals.

These goals can be achieved by designing and carrying out experiments in which live voice experiments are conducted using a real-time simulation of the proposed design.

Phase III. Feasibility Demonstration Experiment

The goals of this phase are:

(a) To design a feasibility demonstration field experiment for the preferred system design using an actual earth terminal/satellite configuration.

(b) To conduct and evaluate such an experiment.

If Phase I and II have been carried out successfully the algorithm, per se, should already be debugged. The Phase III problems will highlight

the problems of interfacing hardware and software with the existing/ available earth terminal equipment and communication network.

9.1.3 Hybrid System Development

The phased approach described in the previous subsection concentrates on developing the basic DVRA voice packet switching concept. As the DVRA concept is developed and if it proves to be as feasible and efficient as it now primises, it is also important for the evolving DCS to develop the full hybrid packet/bit stream concept as described in Chapter 7.

Beginning with the results of Phase I and incorporating results of Phase II and III as they are available, this development should concentrate on the approaches and algorithms to insure:

- Consistency of priority treatment between packet- and circuit-switched traffic,
- Reallocation of capacity between circuit and packet channel fractions (X and Y bits) per earth terminal, and
- Reallocation of capacity between earth terminals.

The last issue above (i.e., inter-terminal capacity reallocation) is in our recommended TDMA implementation equivalent to the earth terminal power reallocation to be performed by the RTAC (Real Time Adaptive Control) system in the near term DSCS-III operation. RTAC however is designed to reallocate earth terminal power in an FDMA implementation of what in this report is called FAMA (Fixed Assignment Multiple Access), and described in Section 4.3.1. With RTAC, the "fixed" assignments are not permanent but can be changed for network rearrangment but not on a call-to-call basis.

An important consideration in developing the hybrid approach, or even just the pure DVRA approach (which also needs the inter-terminal capacity reallocation), is to determine what elements of the RTAC structure can be incorporated.

9.2 VOICE DIGITIZATION STUDIES

The need for further study of packetized voice effects has been described in the previous section in terms of the development of the destination variable random access (DVRA) approach. The present study has brought to light a number of issues where further information is needed to clarify the subjective effects on human communication activities of such parameters (and their interaction) as:

- Packet loss rate
- Packet loss "randomness"
- Delay (fixed)
- Delay jitter
- Type of speech digitization
- Digitization rate.

This kind of information is needed by the designer of any packet voice communication network. The present study has, because of the satellite/RMA problems, tended to emphasize the need for deeper understanding of delay and packet loss effects. The ability of a particular system study to uncover areas of speech research, which need exploring, is exactly why we feel that it is highly productive to integrate that research with analysis/design of particular kinds of communication systems -- as opposed to doing open-ended broad basic research in "digitized speech".

Other areas of packetized speech which, as a result of this study, we feel should be emphasized are:

- Strategies of lost packet smoothing/reconstruction
- Modeling of the speech activity/packet generation

Finally, we remark that on the basis of the experience with the speech experiment in this project, as well as other experiences, that future speech experiments should give more emphasis to a variety of communication task and user acceptance performance measures -- as opposed to standard one-way communication oriented measures such as DRT (intel-legibility).

9.3 EARTH TERMINAL AUTOMATION STUDY

9.3.1 The Need for Earth Terminal Automation

As presented in Chapter 6, the overall system costs of the satellite communications systems as discussed will be dominated by the cost of operation and maintenance (O&M) unless the required manning level can be greatly reduced or eliminated. To illustrate, Figure 9.3.1 shows the potential annual manning cost savings using automated (unattended) earth terminals. For example, at 200 earth terminals, a potential annual savings of $182 million may be realized using unattended earth terminals as opposed to ETs requiring 20 man-levels. Although the results of this figure oversimplify other issues, they do illustrate the attractiveness of unattended or minimally attended ETs as an effective means of reducing the overall system costs of satellite communications. According to a recent projection, DoD expenditures for military communications systems and equipments for FY 1976 through 1985 will total $8.4 billion with voice communications totaling $4.4 billion and data toaling $3.9 billion (9.3-1). If a large number of unattended earth terminals can be deployed to replace some terrestrial trunk lines, overall system costs of communications can be effectively reduced.

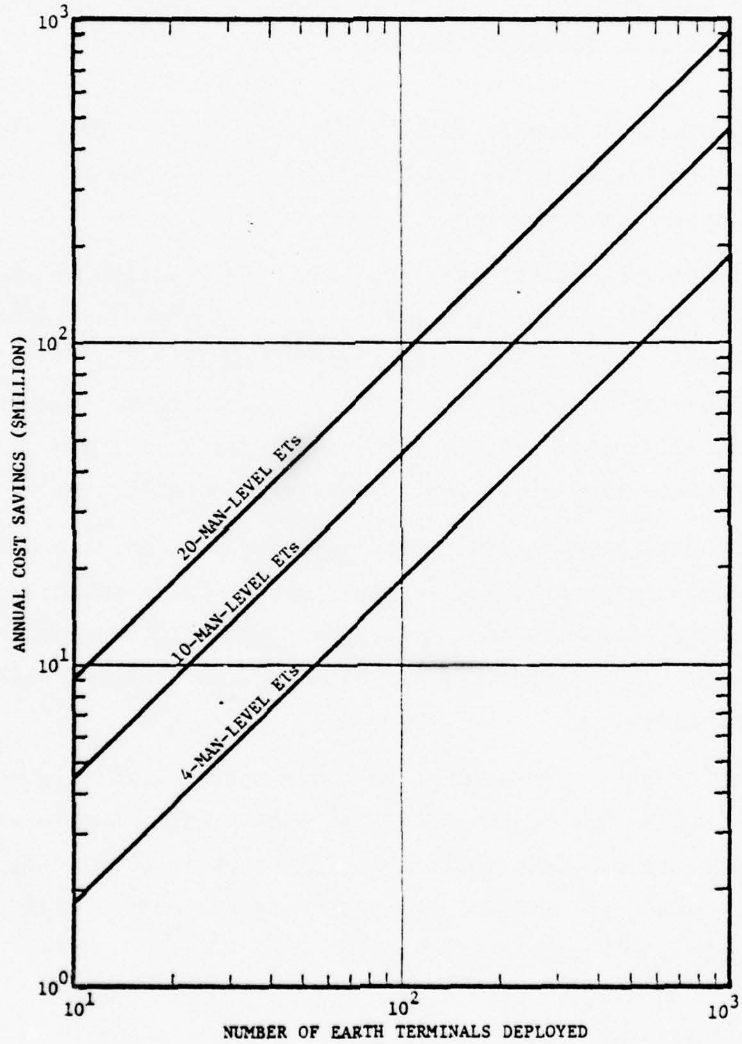


FIGURE 9.3.1 ANNUAL MANNING COST SAVINGS OF AUTOMATED EARTH TERMINALS

9.3.2 Feasibility of Automated ETs

The viability of automated or minimally attended ETs is directly related to the DAMA concept is satellite communications. This happens in a number of ways as discussed below:

- DAMA concepts usually imply a larger number of ETs; the cost of operating these manned ETs render the system not cost effective, in general.
- DAMA concepts usually imply a large congregation of ETs with small traffic loading, which means smaller or less demanding components may be used. As illustrated in Table 9.3.1, these components, generally, have lower failure rates (higher reliabilities). This feature encourages unattended ET operation with preventive or optimally scheduled maintenance.
- Technology advances in space segments, such as more powerful transponders and higher antenna gain in earth coverage or multiple spot-beam configurations, facilitate the deployment of DAMA ETs in large numbers with smaller size and higher reliability.
- In addition to the manning cost savings, the large number of smaller ETs can be mass produced with smaller and modularized components to yield further cost savings in acquisition, operation, and maintenance (including the cost of packaging, shipping, and warehousing).

9.3.3 Proposed Area of Study

The implementation of unattended or minimally attended ETs will be dependent on the cost effectiveness of such systems. Several key issues should be resolved at system level:

TABLE 9.3.1 FDMA FAILURE RATES FOR LNA'S, HPA'S, AND ANTENNA SYSTEMS

	FAILURE RATE $\lambda \times 10^{-6}$ HRS.
LNA's	
1.5 dB N.F.	150
3.0 dB N.F.	100
5.0 dB N.F.	60
7.0 dB N.F.	40
HPA's	
0.50 watts	50
1.00 watts	60
5.00 watts	80
10.00 watts	100
50.00 watts	130
100.00 watts	150
250.00 watts	180
ANTENNA	
15 feet	25
10 feet	10
8 feet	5
6 feet	2
4 feet	1

- Survey of components and subsystems market for ET to establish the cost vs. reliability relationship of ETs as a function of some figure-of-merit, such as G/T or downlink margin.
- Identify viable candidate technical approaches for unattended operation using concepts such as maintaining a hot spare, switched redundancy, preventive maintenance, and optimally scheduled maintenance.
- Make life-cycle cost analysis versus performance trades among candidate approaches to derive a range of recommended approaches.

9.4 IMPACT OF ADVANCED SPACE SEGMENT CONCEPTS

9.4.1 Advanced Space Segment Concepts

As discussed in Section 5.3.2 of this report, advanced concepts such as the regenerative repeater can offer 2-6 dB system gain without any other processing. This amounts to a direct increase of DAMA capability in the system. Furthermore, satellite on-board processing of signals, such as the technique of satellite-switched multiple beams, permit the DAMA concepts implementation an additional degree-of-freedom in design as compared to the one-satellite, non-processing configuration. For example, the concept of beam switching allows frequency reuse as well as spatial demand assignment to take place. The concept of store-and-forward allows a shift in data transmission rate between the uplink and downlink and, thus, facilitates the mixture of large and small ETs within the DAMA network, a desirable feature for a DCS network. Also, RMA techniques, such as one using a capture effect, can be implemented with on-board processing satellites.

In view of these potentials, it is believed that the system-wide advantage of DAMA techniques will be more fully demonstrated if the

impact of regenerative/processing satellites is assessed in this regard. To this end, an immediate objective would be the assessment of a number of multiple-access concepts, in terms of their cost/performance characteristics, in conjunction with a regenerative/processing satellite. Particularly, several specific tasks are proposed (in the section below) as essential to the continuation of this study.

9.4.2 Proposed Area of Study

The following tasks are proposed to evaluate the cost effectiveness of DAMA techniques as applied to regenerative/processing satellites. A DCS common-user environment identical or similar to that of this study will be assumed. These tasks do not involve any foreseeable technical risk in the system study level.

- Reassessment of annual satellite/ET system cost characteristics of DAMA techniques substituting a regenerative repeater and processing satellite for the translating repeater assumed in this study. The anticipated end result of this task will be the recommendation of a DAMA/satellite configuration (e.g., processing or non-processing) for a given DCS common-user loading. However, the methodology developed and the associated parametric results will also be useful for subsequent system developments.
- Development of system analyses of various RMA concepts using a process satellite. This task calls for the mathematical modeling and analysis of system throughputs of various RMA concepts that are processing satellite peculiar.
- System analysis of configurations using satellite-switched TDMA, satellite-switched-regenerative TDMA, and other hybrid multiple-accessing schemes as applied to the DCS common-user model. Should the results prove favorable, hardware laboratory experiments and simulations can be proposed to determine key implementation issues.

9.5 OTHER AREAS OF ADDITIONAL RESEARCH

9.5.1 Cost Trades Between HPA for TDMA and RMA and the Circuitry Complexity for Spread-Spectrum RMA

The results of this study indicate that SS-RMA does not offer a system gain advantage over the regular RMA. However, the cost of high peak-power HPA required for TDMA or RMA operation may be offset by the cost of SS-RMA circuitry complexity, which may be decreasing in view of recent microprocessor development. SS-RMA does offer additional A-J and RFI protection in dense terminal areas. As a consequence, it is desirable to develop cost comparisons between these systems on an equal performance basis. The proposed tasks would involve:

- Detailed vendor survey of CW and pulsed HPA cost
- Detailed performance/cost characterization of the regular RMA and SS-RMA in DCS user environments.

9.5.2 Bandwidth Efficient Modulation

If appears that higher levels of PSK, APM (amplitude/phase modulation), MSK (minimum shift-keyed modulation), and other novel modulation techniques may be suitable for digital/data and voice transmission. Their applicability to SENET-type concepts should be addressed. This task is perhaps out of the scope of a study of this kind, but their effects should be studied further.

REFERENCES

1 Abramson, N., "The ALOHA System - Another Alternative for Computer Communications," ALOHA System Technical Report B70-1, University of Hawaii, April 1970; proceedings of AFIPS 1970 Fall Joint Computer Conference, Vol. 37, pp. 281-285.

2 Abramson, N., "Satellite Packet Broadcasting to Very Small Earth Stations," CDEC Technical Note #38-75, September 1975.

3 Abramson, N., "The ALOHA System," in Computer-Communications Networks, Norman Abramson and Franklin F. Kuo, Editors, Prentice-Hall, 1973.

4 Abramson, N., "Packet Switching with Satellites," National Computer Conference, New York. June 4-8, Vol. 42, pp. 695-702, 1973.

5 Becker, R., "Assessment of Acceptability of Digital Speech Communication Systems," PRI Annual Report, May 1975, ARPA Contract #DAHC04-75-C-0008.

6 Berlekamp, W. E., "Algebraic Coding Theory," McGraw Hill, 1968.

7 Binder, R., et al., "ALOHA Packet Broadcasting - A Retrospect," University of Hawaii Technical Report No. B75-4, January 1975.

8 Binder, R., "A Dynamic Packet Switching System for Satellite Broadcast Channels," ALOHA System Technical Report B74-5, University of Hawaii, August 1974; International Conference on Communications, Conference Record, Vol. III, San Francisco, California, June 1975.

9 Binder, R., McQuillan, J. M. and R. D. Rettberg, "The Impact of Multi-Access Satellites on Packet Switching Networks," Proceedings of EASCON IEEE 1975, p. 63-A.

10 Boag, J. F., "Some Considerations in the Design and Operation of a Demand Assignment Signalling and Switching Sub-System (DASS)," International Conference on Digital Satellite Communication, November 25-27, 1969, London, England.

11 Bortels, W. H., "Simulation of Interference of Packets in the ALOHA Time-Sharing System," ALOHA System Technical Report B70-2, University of Hawaii, March 1970.

12 Brady, P. T., Effects of Transmission Delay on Conversational Behavior on Echo-Free Telephone Circuits," BSTJ, Vol. 50, No. 1, January 1971, pp. 115-134.

13 Carleial, A. and M. Hellman, "Bistable Behavior of ALOHA-Type Systems," IEEE Trans. on Communications, Vol. COM-23, No. 4, April 1975.

14 Chiao, J. T. and F. Chethik, "Satellite Regenerative Repeater Study," IEEE Canadian Communication and Power Conference, 1976.

15 Cosell, D. and J. Makhoul, "Variable Wordlength Encoding," presented at the 88th meeting of the Acoustical Society of America, St. Louis, November 7-10, 1974.

16 Crowther, W., et al., "A System for Broadcast Communications: Reservation ALOHA," Proceedings of the Sixth Hawaii International Conference on Systems Sciences, January 1973.

17 Davies, R. S., Private Communication.

18 Davies, R. S., "A Survey of Advanced Communication Satellite Techniques," Philco-Ford Corporation (For Aerospace & Communications Corporation) memo, October 28, 1974.

19 Dicks, J. L. and M. P. Brown, Jr., "FDMA for Satellite Comminication Systems," EASCON-74.

20 Forgie, J., "Speech Transmission in Packet-Switched Store-and-Forward Networks," National Computer Conference, 1975, pp. 137-142.

21 Forgie, J., Private Communication.

22 Fralick, S. and J. Garrett, "Technological Considerations for Packet Radio Networks," Proceedings of National Computer Conference 1975, Anaheim, California, p. 233.

23 Fralick, S. and J. Leung, "Study of Throughput and Delay of Spread-Spectrum Multiple Access Modes," Packet Radio Note #2, Stanford Research Institute, Menlo Park, California, April 1974.

24 Haberle, H., Herter, E. and G. Schmidt, "TDMA and Switching," International Conference on Digital Satellite Communication, November 25-27, 1969, London, England.

25 Hadfield, B. M., "Satellite Systems Cost Estimation," Trans. IEEE, Vol. COM-22, No. 10, October 1974.

26 Hanni, M., Ostermann, B. and H. Rupp, "Possibilities of Increasing the Efficiency of TDMA Systems," International Conference on Digital Satellite Communications, November 25-27, 1969, London, England.

27 Hellman, M., Private Communication.

28 Hilborn, G., "Random Multiple Access Satellite Packet Switching by Selective Channel Capture," Proceedings of International Conf. on Information Sciences and Systems, August 19-24, 1976, Patras, Greece.

29 Huggins, "Effect of Lost Packets on Speech Intelligibility," NSC Note #78, February 1976.

30a Hwa, H. R., "A Simulation Study of Packet Switching in Conflict-Free Multi-Access Broadcast Data Communications Systems," Sydney/ALOHA Working Paper 7, January 1976.

31 Justeson, J., "On the Complexity of Decoding Reed-Soloman Codes," IEEE Trans. on Information Theory, Vol. IT-]], pp. 237-238, 1976.

32 Kleinrock, L. and S. S. Lam, "Dynamic Control Schemes for a Packet Switched Multi-Access Broadcast Channel," NCC, 1975.

33 Kleinrock, L. and S. S. Lam, "Packet Switching in a Multi-Access Broadcast Channel: Dynamic Control Procedures," IEEE Trans. on Communications, Vol. COM-23, No. 9, September 1975.

34 Kleinrock, L. and S. S. Lam, "Packet Switching in a Multi-Access Broadcast Channel: Performance Evaluation," IEEE Trans. on Communications, Vol. COM-23, No. 4, April 1975; IEEE Trans. on Information Theory, Vol. IT-22, pp. 410-422, 1976.

35 Kleinrock, L. and S. S. Lam, "Packet Switching in a Slotted Satellite Channle," National Computer Conference, New York, Vol. 42, pp. 703-710, 1973.

36 Kleinrock, L. and S. S. Lam, "Packet Switching in a Slotted ALOHA Channle," AFIPS Conference Proceedings, Vol. 42, 1973.

37 Kleinrock, L. and W. Noylor, "On Measured Behavior of the ARPA Network," Proceedings AFIPS 1974 NCC, Vol. 43, pp. 767-780.

38 Kleinrock, L., Noylor, W. and H. Opderbeck, "A Study of Line Overhead in the ARPANET," Communications of the ACM, Vol. 19, No. 1, pp. 3-13, January 1976.

39 Kleinrock, L. and F. Tobagi, "Packet Switching in Radio Channels: Part I - Carrier Sense Multiple-Access Modes and Their Throughput Delay Characteristics," IEEE Trans. on Communications, Vol. COM-23, No. 12, December 1975.

40 Kleinrock, L. and F. Tobagi, "Packet Switching in Radio Channels: Part II - The Hidden Terminal Problem in Carrier Sense Multiple-Access and the Busy Tone Solution," IEEE Trans. on Communications, Vol. COM-23, No. 12, December 1975.

41 Kleinrock, L. and F. Tobagi, "Random Access Techniques for Data Transmission over Packet Switched Radio Channels," NCC, 1975.

42 Koga, K., Muratani, R. and A. Ogawa, "On a Satellite Regenerative Repeating System" IEEE Canadian Communication and Power Conference, 1976.

43 Konheim, A. and B. Meister, "Waiting Lines and Times in a System with Polling," Journal of the Association for Computer Machinery, Vol. 21, No. 3, July 1974.

44 Lake, O. L., "Quality and Intelligibility Performance of Narrowband Digital Voice Terminals," DCA Internal Memo, June 1976.

45 Lam, S. S., "Packet Switching in a Multi-Access Broadcase Channel with Application to Satellite Communication in a Computer Network," Ph.D. Dissertation, Dept. of Computer Science, University of California, Los Angeles, March 1974.

46 Lindsey, W. C. and M. K. Simon, "Telecommunication Systems Engineering," Prentice-Hall, 1973.

47 Magill, D. T., "Adaptive Speech Compression for Packet Communication Systems," Telecommunications Conference Record, IEEE Publ. #73, CH0805-2 29D 1-5.

47a Metzner, "On Improving Utilization in ALOHA Networks," IEEE Transactions on Communications, Vol. COM-24, No. 4, April 1976, pp. 447-448.

48 Mitchell, W. C., Private Communication.

49 Noll, P., "A Comparative Study of Various Quantization Schemes for Speech Encoding," Bell System Tech. J., Vol. 54, pp. 1597-1614, 1975.

50 Peterson, W. W., "Error Correcting Codes," MIT Press, 1961.

51 Purton, R. F., "A Comparison of Two Digital Systems for Demand Assignment of a Satellite Circuit," Proceedings of the INTELSAT/IEEE International Conference on Digital Satellite Communication, November 25-27, 1969, London, England, p. 355.

52 Puente, J. G., et al., "Multiple-Access Techniques for Commercial Satellites," Proceedings of the IEEE,Vol. 59, No. 2, February 1971, pp. 218-229.

53 Rettberg, R., "Preliminary Simulation Results for Reservation ALOHA," ARPA Network Information Center, Stanford Research Institute, Menlo Park, California, Note 43, NIC Document 16086, May 1973.

54 Rettberg, R., "Random ALOHA with Slots-Excess Capacity," ARPANET Satellite System Note 18, NIC Document 11865, October 11, 1972.

55 Roberts, L. G., "ALOHA Packet System With and Without Slots and Capture," ARPA Network Information Center, Stanford Research Institute, Menlo Park, California, ASS Note 8, NIC 11290, June 1972.

56 Roberts, L. G., "ARPANET Satellite System," Notes 8 (NIC Document 11290) and 9 (NIC Document 11291).

57 Roberts, L. G., "Dynamic Allocation of Satellite Capacity Through Packet Reservation," Proceedings of the National Computer Conference, pp. 711-716, 1973.

58 Roberts, L. G., "Network Rationale: A 5-year Reevaluation," Proc. COMPCON 73, February 1973.

59 Rosner, R. D., "Optimization of the Number of Ground Stations in a Domestic Satellite System," Proc. EASCON 75, pp. G4-A-I.

60 Rosner, R. D., "An Advanced Communications System Concept Employing Demand Assignment Satellites," DCDC TEchnical Note No. 15-75.

61 Rusch, R. J. and D. G. Dwyre, "INTELSAT V Spacecraft Design," Proc. 27th International Astronautical Congress, Anaheim, California, 1976.

62 Schmidt, W. G., "Satellite TDMA Systems" Telecommunications, August 1973.

63 Schmidt, W. G., Gabbard, O. G., Cacciamani, E. R., Maillet, W. G., and W. W. Tu, "MAT-1: INTELSAT's Experimental 700-Channel TDMA/DA System," International Conference on Digital Satellite Communication, November 25-27, 1969, London, England.

64 Schwartz, J. W., et al., "Modulation Techniques for Multiple Access to a Hard-Limiting Satellite Repeater," Proc. IEEE, Vol. 54, May 1966.

65 Shannon, C. E., "Communication in the Presence of Noise," Proc. IRE, 37, pp. 10-21, 1949.

66 Smith, J. M. and E. Stern, "Surface Acoustoelectric Convolvers," Proc. 1973 IEEE Ultrasonics Symp., Monterey, California, November 1973, pp. 142-144, IEEE Order 73CH0807-8SU.

67 Stein S. and J. J. Jones, "Modern Communication Principles," McGraw-Hill, New York 1967.

68 Tillotson, L. C., "A Model of a Domestic Satellite Communication System," BSTJ, Vol. 47, No. 10, 1968.

69 Tobagi, F., "Random Access TEchniques for Data Transmissions over Packet Switched Radio Network," School of Engineering and Applied Science, Univerisity of California, Los Angeles, UCLA-ENGR 7499, December 1974.

70 Un, C. K. and D. T. Magill, "The Residual-Excited Linear Vocoder with Transmission Rate Below 9.6 Kbits/s," IEEE Trans. on Communication, Vol. COM-28, No. 12, december 1975.

71 Viterbi, A. J., "Convolutional Codes and Their Performance in Communication Systems," IEEE Trans. on Communication Technology, Vol. COM-19, pp. 751-772, 1971.

72 Voiers, A. D., et al., "Research on Quality Evaluation and Procedures," DCA Contract Report 100-74-C-0056, January 1976.

73 Werth, A. M., "SPADE: A PCM FDMA Demand Assignment System for Satellite Communications," International Conference on Digital Satellite Communication, November 25-27, 1969, London, England; IEEE Interntional Communication Conference, 1970.

APPENDIX A

TWO-WAY COMMUNICATION EVALUATION OF PACKET NETWORK VOICE QUALITY DEGRADATION

APPENDIX A TWO-WAY COMMUNICATION EVALUATION OF PACKET NETWORK VOICE QUALITY DEGRADATION

The hardware used in the two-way conversation experiment is of two types: (1) the voice channel equipment and (2) the user console control and scoring equipment.

(1) The voice channel equipment is based on a PDP 11/40 minicomputer and CVSD voice digitization. Two analog interfaces were built to allow two users to call-in through the SRI PBX (local extensions 2262 and 2162). The analog signals to/from the handset are separated and amplified to proper levels for input/output for the General Atronics MX-521 CVSD units. These units are interfaced to the 11/40 through DR11-Cs (parallel line units). The program in the 11/40 simulates a full duplex transmission channel between the two CVSD units. The types of degradation that can be introudced in this simulated channel are packet delays and packet losses. The packet losses are determined by setting a probability for losing a packet. A random number chosen from a uniform distribution is compared to this probability to determine whether the packet should be discarded. A second conditional probability can be set such that given the last packet was discarded, the next packet may be discarded with a different probability. This Markov loss model allows simulation of complicated transmission channel behavior. Provision has been made for adding introduction of channel bit errors at a later time.

(2) The user console control and scoring equipment is based on a PDP 11/10 computer. Two user consoles are interfaced to the 11/10 through parallel line units. Each console is located in an acoustically quiet room so that background noise can be controlled. An SRI PBX extension phone is in each room, for access to the voice channel equipment. The consoles have

four light-emitting diodes (LEDs) to signal experiment task initiation. Also, there is a four-digit numeric display. Four buttons are available to record user responses. The 11/10 program can record inter-response times, present various digital patterns, and record the user response. Various other psychoacoustic and physiological measurements can also be made for expanded experiments.

The task chosen for the pilot experiment is a simple number proof-reading task. In this task one subject is seated in a quiet room A; the other subject is seated in quiet room B and uses a standard Western Electric telephone to communicate with the first subject over the simulated packet speech network. A four-digit number is displayed to both subjects. This number may be the same for both subjects or it may be the same except for one digit. In this experiment, the probability of the numbers being the same is set to 0.5, so that it is equally likely that the numbers will be the same or be different. The LEDS are used to indicate which subject is to initiate conversation. The subjects must communicate with each other to determine whether they are both looking at the same number or at different numbers.

If they determine that they are looking at the same number, the initiating subject presses button 1, and the other subject presses button 4. If they determine that the two numbers are different, they each press the button corresponding to the digit that is in error, i.e., if subject A has the number 7602 and subject B has the number 7402, both should press button 2. If both subjects made the correct response, the displays are blanked for 2 seconds and then a new set of numbers is displayed. The times required for each subject to respond correctly are stored for later analysis. If one or both subjects do not respond correctly, the displays do not blank. In this case the subjects must communicate to determine what mistake has been made, and again each must push the correct button in order to blank the display and go on to the next set of numbers.

Because of current limitations in input/output capability, the 4 digit displays are restricted to digits from 0 to 7. Thus, when the numbers are the same, 12 bits of information must be successfully transmitted from one subject to the other, and 1 bit of information (that the numbers are the same) must be transmitted back. When the numbers differ, twelve bits of information must be successfully transmittted one way and two bits of information (the location of the differing digit) must be transmitted back. By measuring the time required to successfully complete each number proof-reading task, we can measure the effective rate of information exchange as a function of the communication system parameters.

In the pilot experiment, nine different conditions were studied. One of the conditions was a control good condition consisting of 16 kbits/s CVSC speech transmitted in 20 ms packets with a loss rate of 1% and no delay in the channel. At the other extreme, a control bad condition consisted of 8-kbits/s CVSD speech transmitted in 100 ms packets with a 10% loss rate and 540 ms one-way delay in the channel.

Three conditions were used to investigate the effects of fixed channel delay. In these conditions 16-kbits/s speech in 100 ms packets with a 1% loss rate; fixed one-way delays of 270, 540, and 1080 ms were used. Two conditions investigated packet loss rates of 1% versus 10% using 16-kbits/s speech in 100 ms packets with a fixed delay of 270 ms. Packet length was varied in three conditions, with durations of 20, 50, and 100 ms. For these conditions 16-kbits/s speech was transmitted with a 10% loss rate and a 270 ms fixed delay. Finally, the effect of digitization rate was investigated using 16-kbits/s and 8-kbits/s speech, with 100 ms packets, 10% loss rate, and a 270 ms fixed delay.

Training was accomplished in three sessions prior to the experimental conditions. These sessions used the control good condition, the

control bad condition, and the condition that would be the last condition to be heard by that pair of subjects. Because it was likely that subjects would not be completely trained after three sessions and would continue to improve during the experimental session, the conditions were rotated in time for the different subject pairs. That is, if condition one occurred first for subject pair one, it occurred second for subject pair two, third for subject pair three, and so on.

APPENDIX B

SATELLITE COMMUNICATION SYSTEM

COST ESTIMATION

B.1 COST

B.1.1 Cost Analysis Methodology

To facilitate the subsequent global analyses of cost and saving of replacing terrestrial communication networks with satellite communication systems, using DAMA techniques where the deployment of a large number of earth terminals is made to replace the access network as well as the backbone trunking, a cost model is proposed as illustrated in sequence of Figure B.1.1, B.1.2 and B.1.3. This section is concerned only with the assessment of curve A (annual earth terminal cost) and curve B (annual satellite cost), and the local optimization of the two.

B.1.2 Earth Terminal Cost

The major components of the earth terminal in terms of the present technological approach are the following:

- Voice processing unit/modem
- Up/down converters
- High-power amplifier (HPA)
- Low-noise amplifier (LNA)
- Antenna
- Access control computer

of which the access control computer is highly dependent of the demand-access technique involved and will not be considered at the present time.

To allow subsequent intra-terminal as well as ET and satellite cost tradeoff analyses, component-cost characteristics must be established. Table B.1 shows the respective characteristics to be collected for each component, with its potential suppliers as data base.

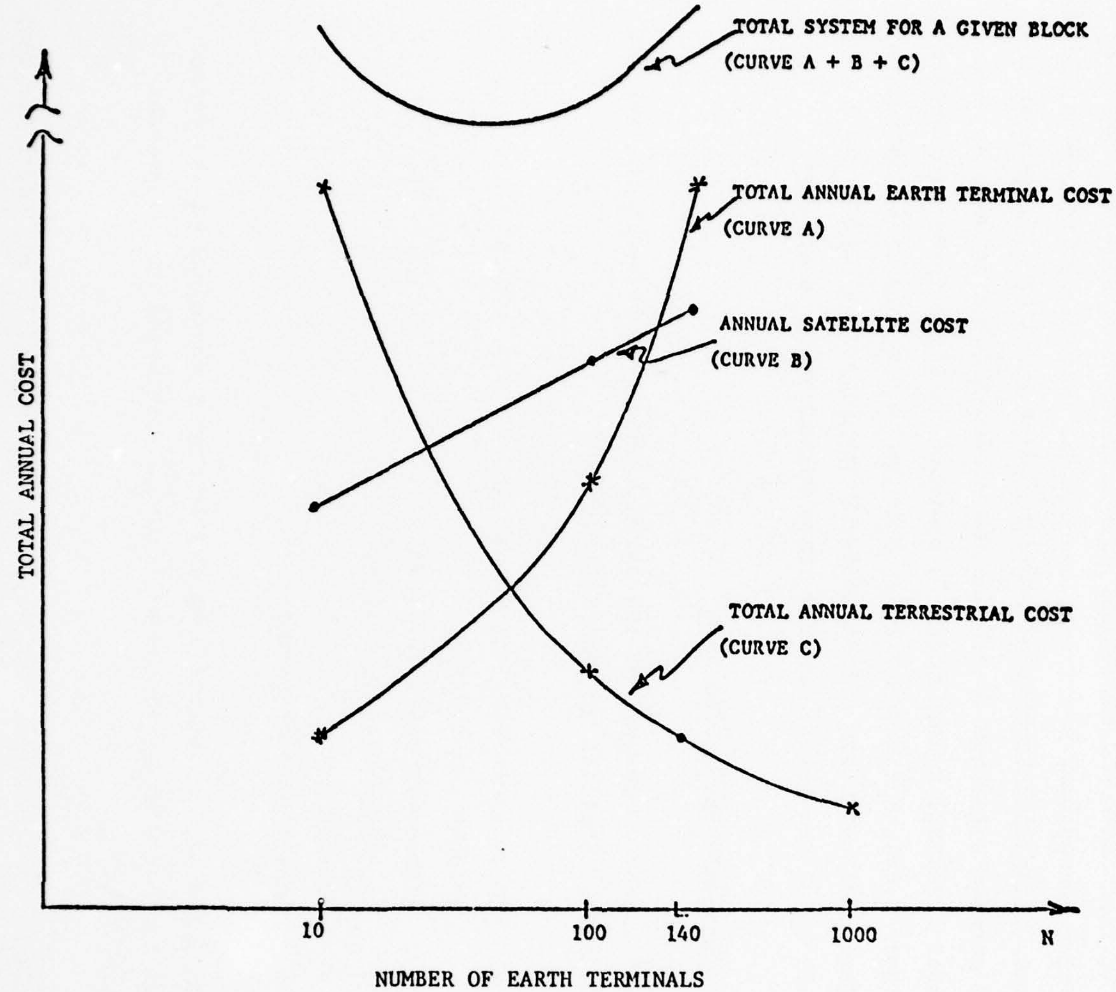


FIGURE B.1.1 TOTAL ANNUAL SYSTEM COST AS A FUNCTION OF NUMBER OF EARTH TERMINALS FOR A CANDIDATE SYSTEM

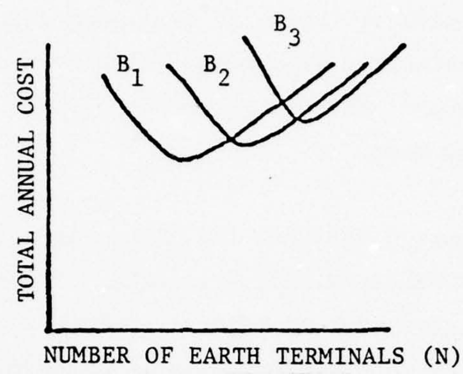


FIGURE B.1.2 TOTAL ANNUAL CANDIDATE SYSTEM COST AS A FUNCTION OF NUMBER OF EARTH TERMINALS FOR DIFFERENT BLOCKING

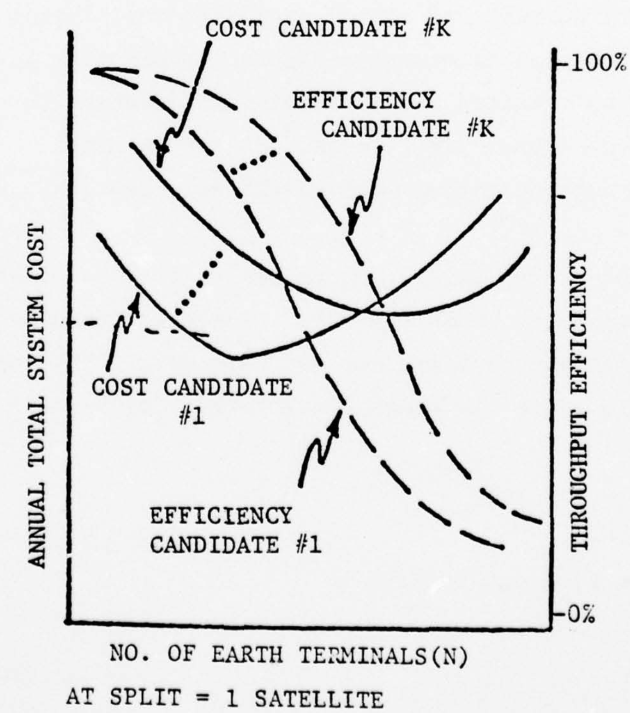


FIGURE B.1.3 COMPARISON OF CANDIDATE SYSTEMS AT VARIOUS EARTH/SATELLITE LOAD SPLITS

The budgetary prices of the major RFI equipment items of a satellite earth terminal were solicited from representative equipment manufacturers. Price quotes and equipment data sheets were obtained from at least two manufacturers for each item.

The price information obtained for LNAs, PAs and antennas has been summarized in Figures B.1.4, B.1.5, B.1.6 and B.1.7 respecitvely. As was expected, C-band equipment is readily available for all parameter values of interest. Ku-band equipment is being produced in fewer "capacity" (size, power, temperature) ranges, but most manufacturers will be increasing the ranges of their Ku-band offerings in the future.

X-band equipment was less readily available in commercial quality. It was estimated that the price of equipment would increase by a factor of 1.5 to 2.0 in going from best commercial grade to full milspecs.

The effect on price of volume purchases varied from item to item. Generally, when human interaction figured prominently in the production process (such as required when screening transistors to obtain low-noise amplifiers), the volume discount was small. In either case, however, it could be approximated rather well by the standard learning curve technique.

"Labor intensive" production processes were associated with a 98% or 99% learning curve while the other processes could be associated with a 90% to 92% curve. That is, the per unit cost (PUC) for purchases of n units is related to the single-item cost (SIC) by

$$PUC = SIC\ (y)^{\log_2 n} \tag{B.1}$$

where y ranges from 0.9 to 0.99.

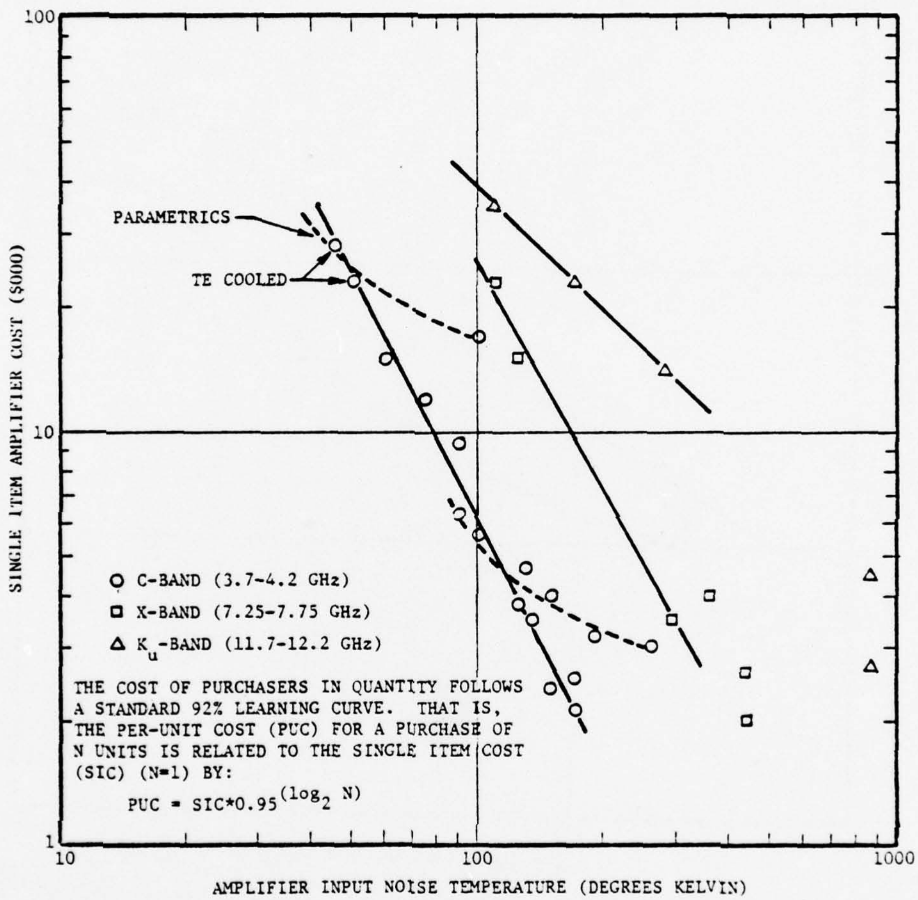


FIGURE B.1.4 LNA COST CHARACTERISTICS

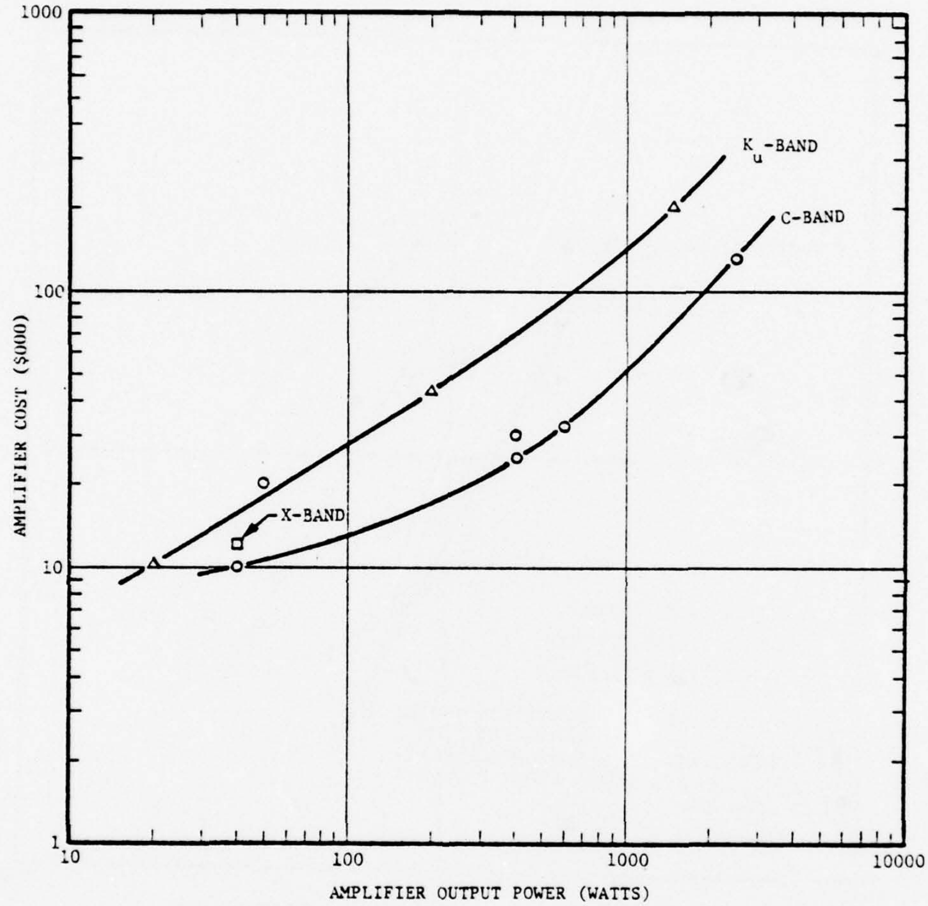


FIGURE B.1.5 HPA COST CHARACTERISTICS

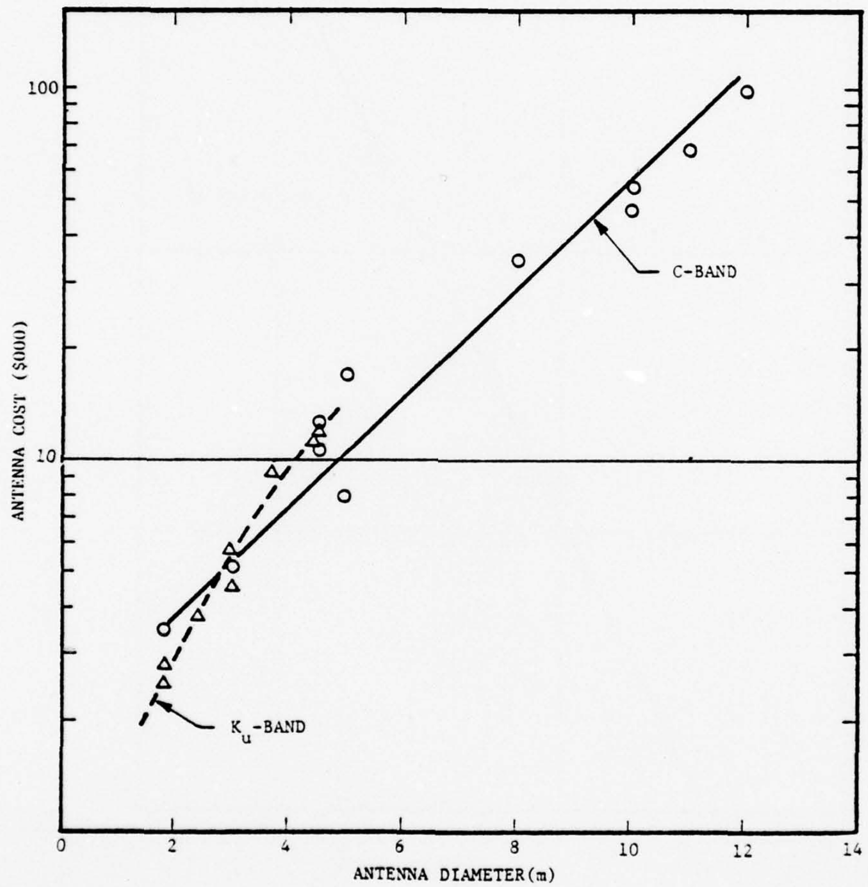


FIGURE B.1.6 ANTENNA COST CHARACTERISTICS

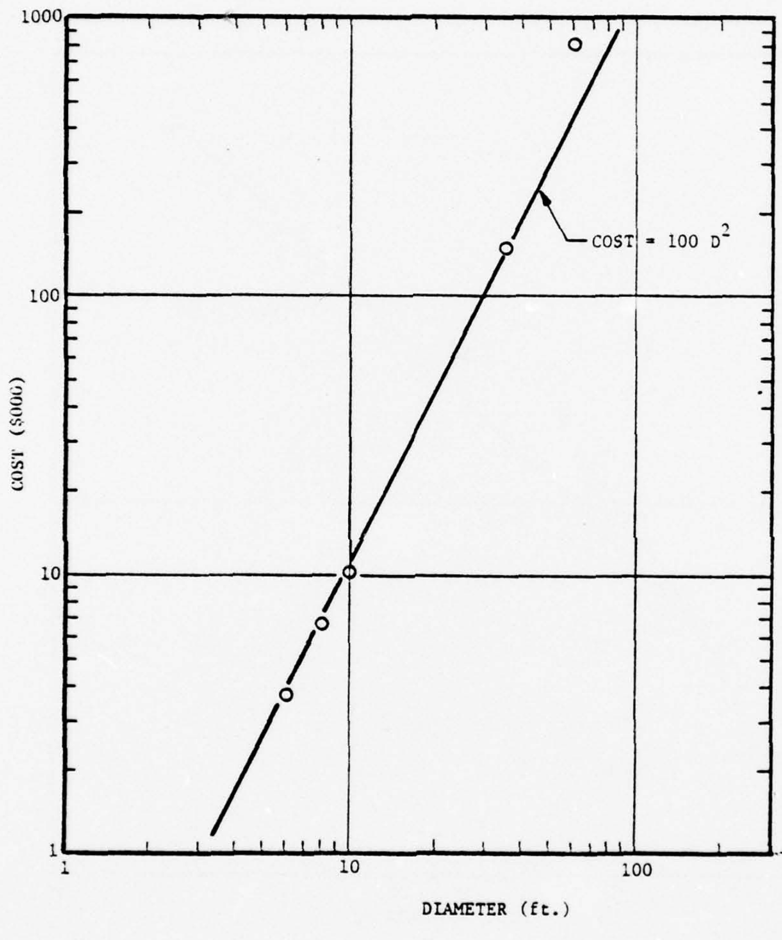


FIGURE B.1.7 X-BAND ANTENNA COST VERSUS DIAMETER

For the larger diameter antennas and the higher power power amplifiers, the demand historically has not provided a basis on which to estimate realistic volume pricing since production runs to be limited to lots of a few, at best. There probably would be some advantage to a positive "bargaining" attitude when firm quotes for quantity purchases are solicited.

To assess the impact of ET sensitivity (G/T) on cost, ET's G/T is plotted as function of single-item cost of LNA and antenna for C, X and Ku-band in Figures B.1.8, B.1.9 and B.1.10, respectively, using the data of Figures B.1.4, B.1.6 and B.1.7. In each of the figures (Figures B.1.8-B.1.10), lower envelope constitutes a minimum cost curve. The figures also suggest that G/T is highly sensitive in cost for military grade X-band and is much less so for commercial grade C-band. In either case, the cost increases monotonically with the increase of G/T.

B.1.3 Space Segment Cost

The space segment cost can be estimated on a buy-or-lease basis.

Assuming that the satellite technology employed is fairly standard so that only the recurring costs are involved, the cost of production and emplacement of a geostationary satellite is given approximately by

$$C_s(\$M) = 0.026\,(W_p)^{2/3}\,\left(1 + K + \frac{22238}{8000}\right)$$

where

W_p is the payload weight in pounds, and K is a constant determined by the payload sophistication. For example, the INTELSAT-V has a payload W_p = 4112.5 lb. Assuming K = 4, the single-satellite cost then is

$$C_s = 0.026(4112.5)^{2/3}\,(1 + 4 + 2.78) = 51.92\text{ M}$$

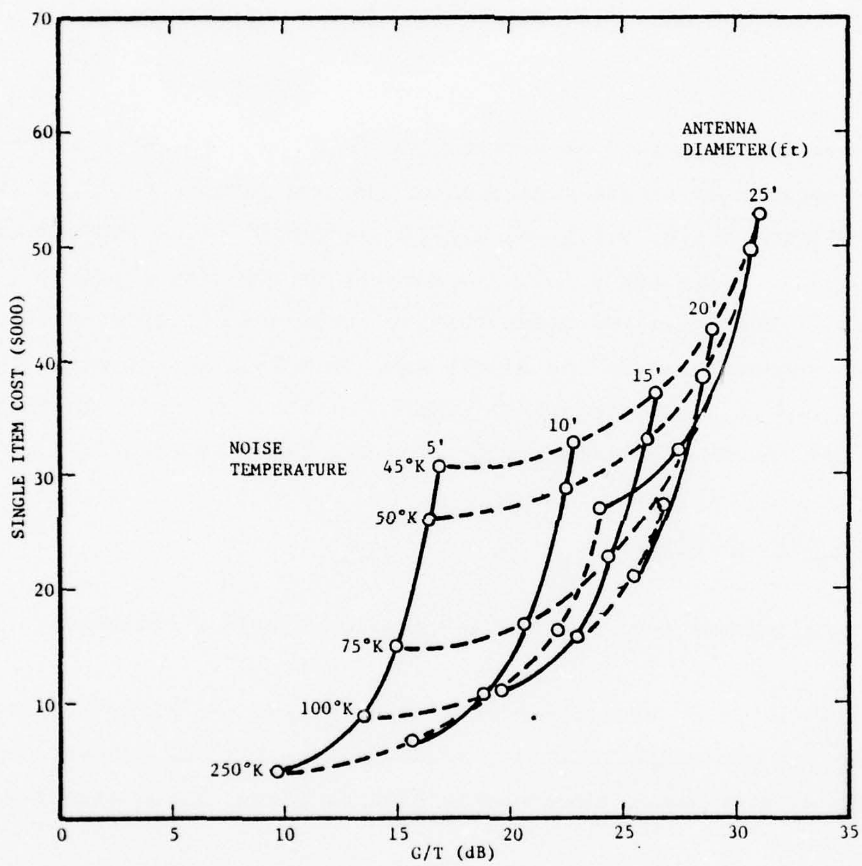


FIGURE B.1.8 C-BAND ET (LNA + ANTENNA) COST

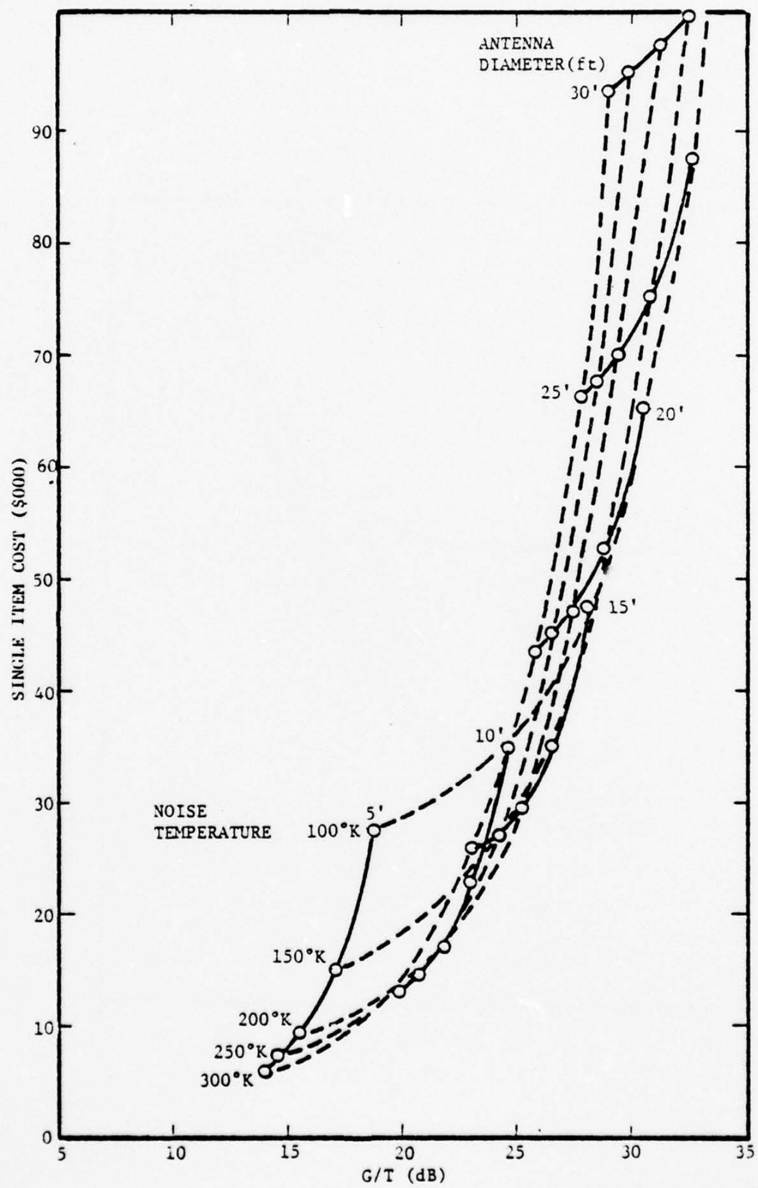


FIGURE B.1.9 X-BAND ET (LNA + ANTENNA) COST

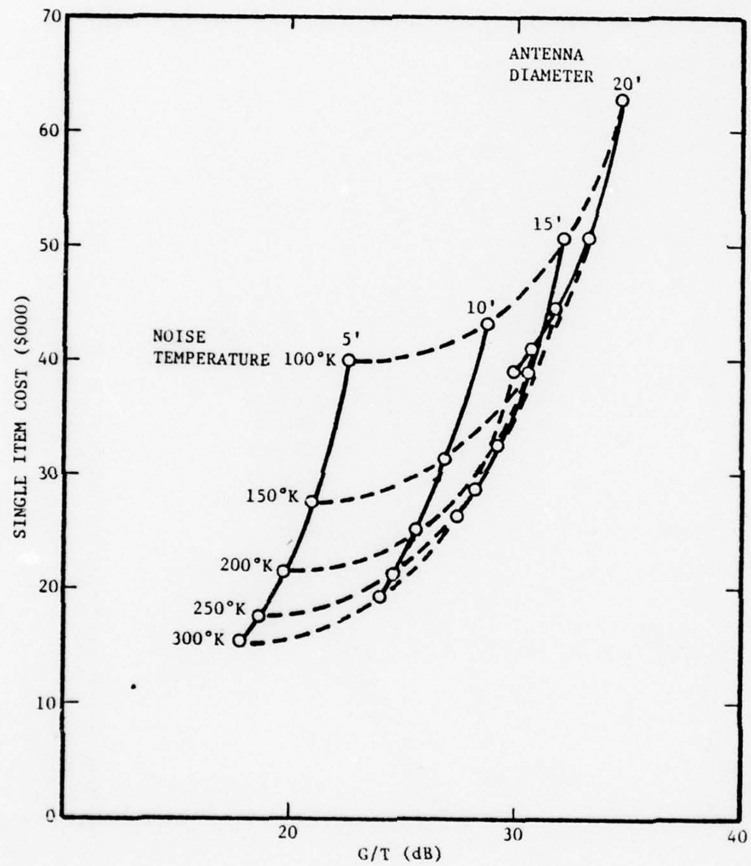


FIGURE B.1.10 K_u-BAND ET (LNA + ANTENNA) COST

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IMPLICATIONS OF DEMAND ASSIGNMENT FOR FUTURE SATELLITE COMMUNIC--ETC(U)
JUN 77 C G HILBORN, A C PAN, P J BOGERT DCA100-76-C-0060

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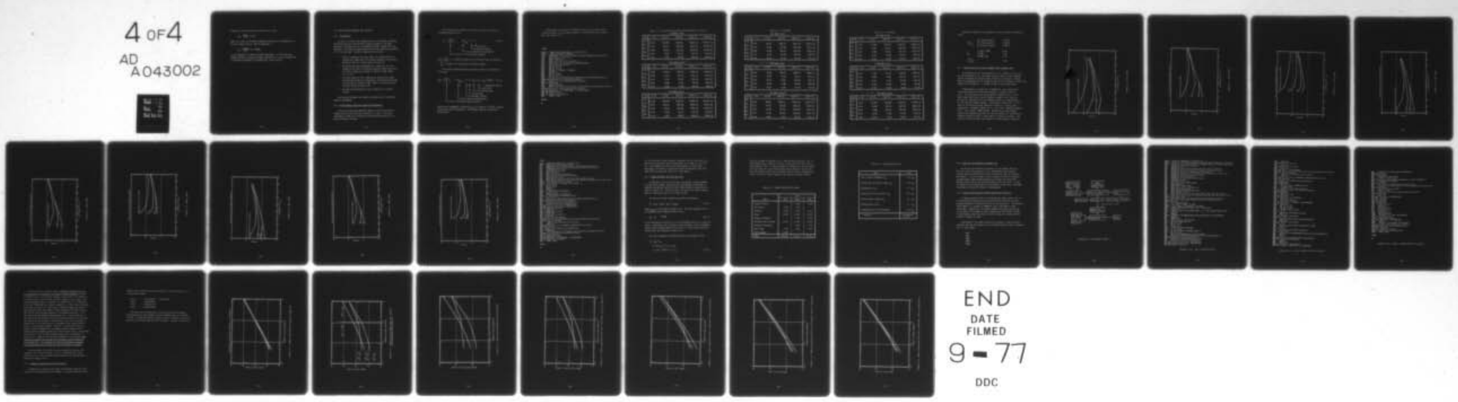

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Assuming a 10-year lifetime, the annual cost is then

$$C_{sa} = \frac{52.92}{6.45} = 8.21 \text{ M}$$

There is a total of 2137 MHz of bandwidth available in an INTELSAT-V, so that the annual cost per 1 MHz of bandwidth is

$$C'_{sa} = \frac{8.21 \text{ M}}{2137} = 38.4 \text{ \$K/MHz}$$

or the equivalent of 1.38M per 36-MHz transponder. On the other hand, INTELSAT generally charges 1M/36-MHz transponder for a spare (preemptible) to 3M/36-MHz transponder (non-preemptible) in bulk leases.

B.2 SATELLITE/EARTH TERMINAL COST ANALYSIS

B.2.1 Introduction

In order to evaluate the system-wide costs on acquiring, operating and maintaining the satellite/earth terminals segment of the DAMA networks, an analysis and the corresponding computer program have been developed. The computer program performs four basic functions applicable to C, X and K_u-bands. Satellite communications systems:

a. Based on components cost data input, the program seeks the minimum cost combination of ET antenna diameter and low noise amplifier (LNA) for a given figure of merit G/T.

b. Determines the power requirement and, therefore, the cost (based on cost data) of high power amplifier (HPA) upon input of traffic loading and the mode of operation (TDMA, FDMA, etc.) at ET.

c. For given number of ET's, computes and compares the annual satellite-ET system cost (acquisition, initialization and O&M) for each mode of operation (TDMA, FDMA, etc.) and identifies the mode having the lowest cost.

d. Provides digital/graphical output capability for computer results.

This section documents the necessary analyses and the associated computer programming.

B.2.2 Earth Terminal High Power Amplifier Requirements

The cost of high power amplifier (HPA) is a function of traffic loading and duty cycle the earth terminal is to carry. The power requirement for HPA can be determined through uplink analyses of the access techniques involved.

The average power required at satellite RF input is given by the following equation in dB.

$$P_R = \left(\frac{E_b}{N_o}\right)_u - (G/T)_{Sat.} + k + R \qquad (B.2.1)$$

Data Rate

Boltzman constant

Satellite receiving G/T

Uplink E_b/N_o required

where $\left(\frac{E_b}{N_o}\right)_u$ is usually assuemd to be 6-dB better than the downlink $\frac{E_b}{N_o}$ to ensure that the system is downlink limited.

The average transmitting power required for HPA at the respective ET is then:

$$P_{HPA} = \left(\frac{E_b}{N_o}\right)_u - (G/T)_{Sat.} + k + R - G_{ET} + L_S + P_{BO} + \text{Margin} \qquad (B.2.2)$$

Transmitter Backoff

Space Loss

ET Antenna Gain

Data Rate

Boltzman Constant

Satellite Receiving G/T

Uplink (E_b/N_o) required

which can be programmed to evaluate P_{HPA} as a function of traffic loading, access modes and RF frequenceis. The program, denoted as DAMA 10 is listed below:

Program DAMA 10 is written in FORTRAN IV and can be run time shared. A simple execution of the program produces the following HPA power requirement table (Table B.2.1):

```
 LIST

100C... DAMA COST ANALYSIS...11-30-76 JTC
110C... PROGRAM NAME---DAMA10
120     DIMENSION D(5),NX(7),PBO(3),GTS(3),GTE(5),PX(3,3,5,7)
130     DATA GTS/-6.,3.,-6./
140     DATA EBNU/13./
150     DATA D/5.,10.,15.,20.,25./
160     DATA NX/1,5,10,100,1000,10000,15625/
170     DATA PBO/10.,3.,0./
300     DO 10 I=1,3
310     DO 20 J=1,3
312     PRINT: ; PRINT: ; PRINT:
320     PRINT:I,J
330     DO 30 K=1,5
340     DO 40 L=1,7
350     PX(I,J,K,L)=EBNU-GTS(J)-228.6+42.01+10.*ALOG10(NX(L))
360&-20.*ALOG10(D(K))+176.17+6+PBO(I)
370     PX(I,J,K,L)=10.**(PX(I,J,K,L)/10.)
380 40 CONTINUE
400     PRINT 50,D(K)
410     PRINT 60,PX(I,J,K,1),PX(I,J,K,2),PX(I,J,K,3),PX(I,J,K,4)
412&,PX(I,J,K,5),PX(I,J,K,6),PX(I,J,K,7)
450 30 CONTINUE ; 20 CONTINUE ; 10 CONTINUE
470 50 FORMAT(F5.1)
480 60 FORMAT(2X,7F10.1)
500     STOP ; END

ready

*
```

TABLE B.2.1 HIGH-POWER AMPLIFIER POWER REQUIREMENTS (IN WATTS)

for SCDC C-Band

G/T	1 CH	10 CH	100 CH	1000 CH	10000 CH
5.	11.48	114.83	1148.31	11483.12	114831.22
10.	2.87	28.71	287.08	2870.78	28707.80
15.	1.28	12.76	127.59	1275.90	12759.02
20.	0.72	7.18	71.77	717.70	7176.95
25.	0.46	4.59	45.93	459.32	4593.25

for SCDC X-Band

G/T	1 CH	10 CH	100 CH	1000 CH	1000 CH
5.	1.45	14.46	144.56	1445.64	14456.39
10.	0.36	3.61	36.14	361.41	3614.10
15.	0.16	1.61	16.06	160.63	1606.27
20.	0.09	0.90	9.04	90.35	903.52
25.	0.06	0.58	5.78	57.83	578.26

for SCDC K_u-Band

G/T	1 CH	10 CH	100 CH	1000 CH	1000 CH
5.	11.48	114.83	1148.31	11483.12	114831.22
10.	2.87	28.71	287.08	2870.78	28707.80
15.	1.28	12.76	127.59	1275.90	12759.02
20.	0.72	7.18	71.77	717.70	7176.95
25.	0.46	4.59	45.93	459.32	4593.25

TABLE B.2.1 (CONTINUED)

for FDMA C-Band

G/T	1 CH	10 CH	100 CH	1000 CH	1000 CH
5.	2.29	22.91	229.12	2291.18	22911.84
10.	0.57	5.73	57.28	572.80	5727.96
15.	0.25	2.55	25.46	254.58	2545.76
20.	0.14	1.43	14.32	143.20	1431.99
25.	0.09	0.92	9.16	91.65	916.47

for FDMA X-Band

G/T	1 CH	10 CH	100 CH	1000 CH	1000 CH
5.	0.29	2.88	28.84	288.44	2884.43
10.	0.07	0.72	7.21	72.11	721.11
15.	0.03	0.32	3.20	32.05	320.49
20.	0.02	0.18	1.80	18.03	180.28
25.	0.01	0.12	1.15	11.54	115.38

for FDMA K_u-Band

G/T	1 CH	10 CH	100 CH	1000 CH	1000 CH
5.	2.29	22.91	229.12	2291.18	22911.84
10.	0.57	5.73	57.28	572.80	5727.96
15.	0.25	2.55	25.46	254.58	2545.76
20.	0.14	1.43	14.32	143.20	1431.99
25.	0.09	0.92	9.16	91.65	916.47

TABLE B.2.1 (CONTINUED)

for TDMA C-Band

G/T	1 CH	10 CH	100 CH	1000 CH	1000 CH
5.	1.15	11.48	114.83	1148.31	11483.12
10.	0.29	2.87	28.71	287.08	2870.78
15.	0.13	1.28	12.76	127.59	1275.90
20.	0.07	0.72	7.18	71.77	717.70
25.	0.05	0.46	4.59	45.93	459.32

for TDMA X-Band

G/T	1 CH	10 CH	100 CH	1000 CH	1000 CH
5.	0.14	1.45	14.46	144.56	1445.64
10.	0.04	0.36	3.61	36.14	361.41
15.	0.02	0.16	1.61	16.06	160.63
20.	0.01	0.09	0.90	9.04	90.35
25.	0.01	0.06	0.58	5.78	57.83

for TDMA K_u-Band

G/T	1 CH	10 CH	100 CH	1000 CH	1000 CH
5.	1.15	11.48	114.83	1148.31	11483.12
10.	0.29	2.87	28.71	287.08	2870.78
15.	0.13	1.28	12.76	127.59	1275.90
20.	0.07	0.72	7.18	71.77	717.70
25.	0.05	0.46	4.59	45.93	459.32

Parameters assumed in the computation of this program are summarized below:

$$(G/T)_{Sat.} : \begin{cases} \text{C-band satellite} & -6\ dB/^{o}K \\ \text{X-band satellite} & +3\ dB/^{o}K \\ K_u\text{-band satellite} & -6\ dB/^{o}K \end{cases}$$

$$P_{BO} : \begin{cases} \text{SCPC - FDMA} & 10\ dB \\ \text{FDMA} & 3\ dB \\ \text{TDMA, RMA} & 0\ dB \end{cases}$$

$$\left(\frac{E_b}{N_o}\right)_u : \qquad 13\ dB$$

B.2.3 Single Item Cost of Earth Terminal--LNA + Antenna + HPA

By combining the cost characteristics of G/T (Figures B.1.8-B.1.10, and of HPA (Figure B.1.5), the single item cost (SIC) of earth terminal portion containing LNA, Antenna dish and HPA can be computed. Program DAMA 12 accomplishes this by integrating the results of Figures B.1.8-B.1.10, Figure B.1.5 and DAMA 10. A listing of DAMA 12 is provided below:

Program DAMA 12, written also in FORTRAN IV, can be executed in time shared system with interactive graphics to produce outputs in plots. The resulting curves are presented in Figures B.2.1 - B.2.9, where single-item costs of earth terminal (antenna + LNA + HPA) are plotted as a function of G/T for terminals capable of transmitting 1, 10, 100 and 1000 16-Kbps channels. Particularly, Figures B.2.1 - B.2.3 refer to the multiple-access technique of SCPC at C, X and K_u-band, respectively. Likewise, Figures B.2.4 - B.2.6 refer to FDMA and Figures B.2.7 - B.2.9 refer to TDMA and RMA. Examination of these figures reveals that minimum cost points are such that higher G/T (25-dB and up) terminals should be used for large number of channels in general. And lower G/T (20-dB and less) are optimal for smaller number of channels.

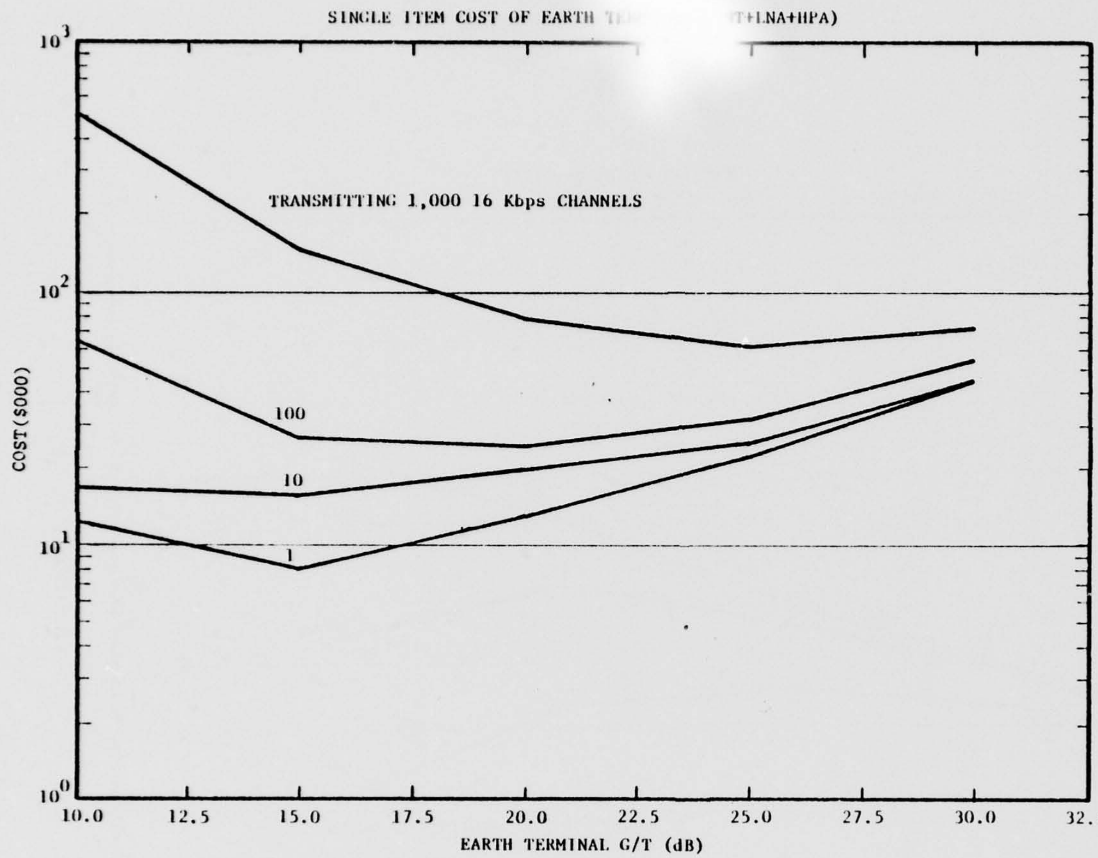

FIGURE B.2.1 SCPC C-BAND

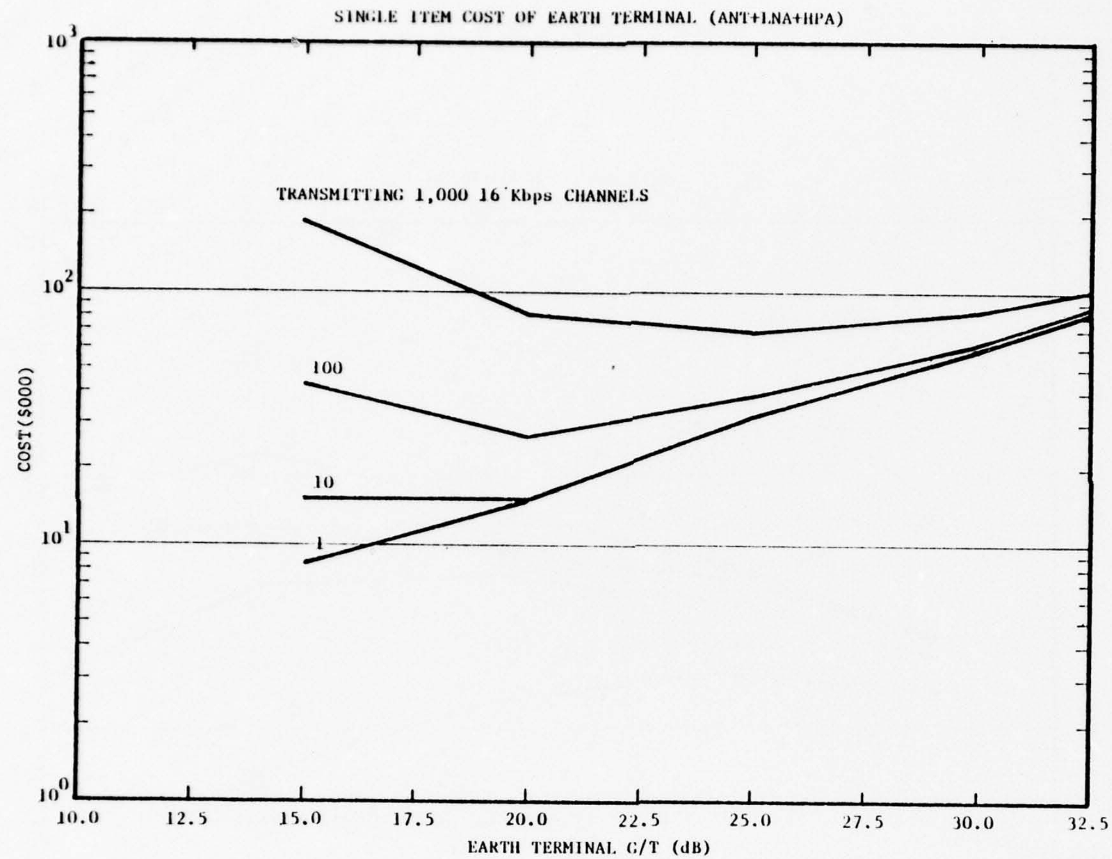

FIGURE B.2.2 SCPC X-BAND

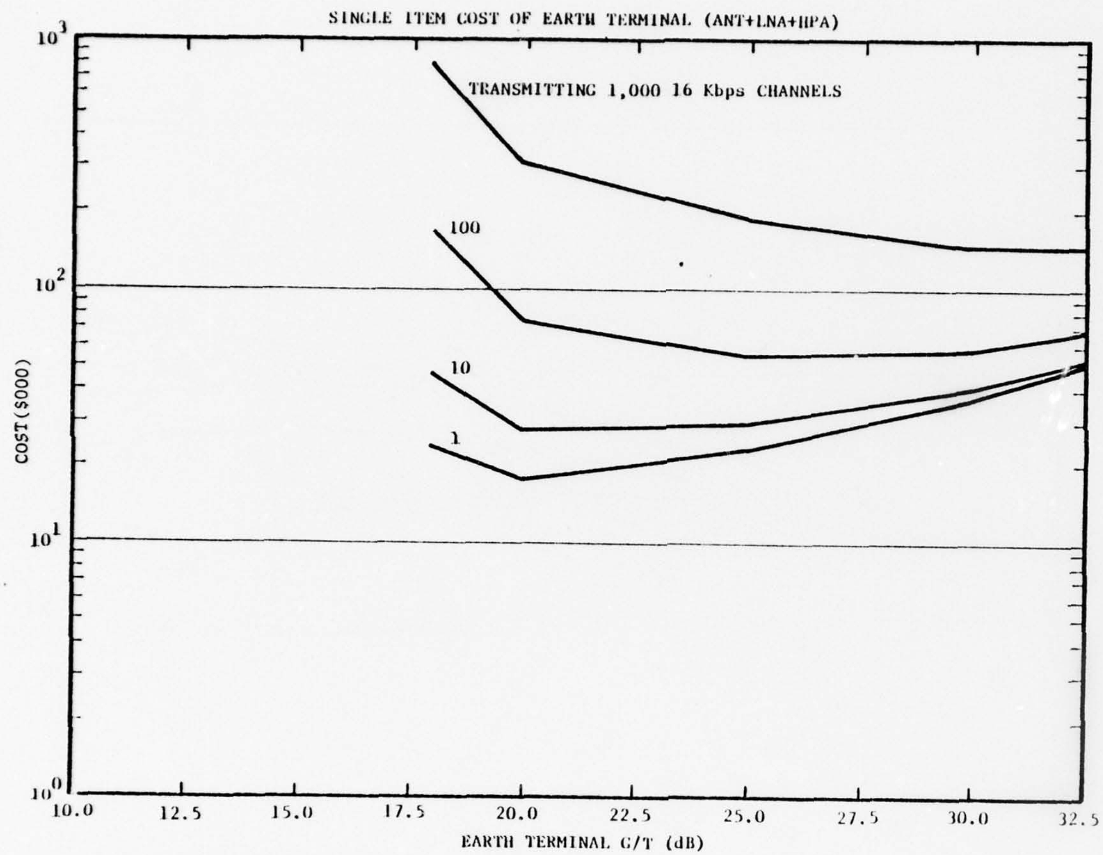

FIGURE B.2.3 SCPC K_u-BAND

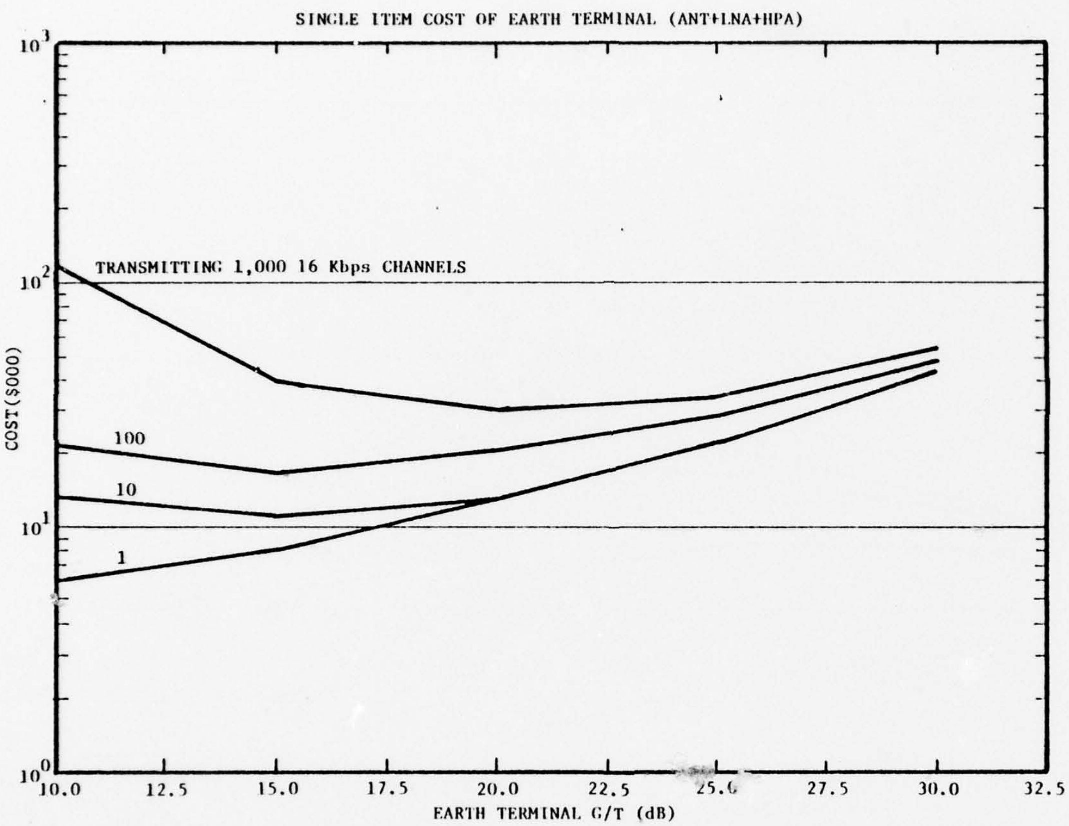

FIGURE B.2.4 FDMA C-BAND

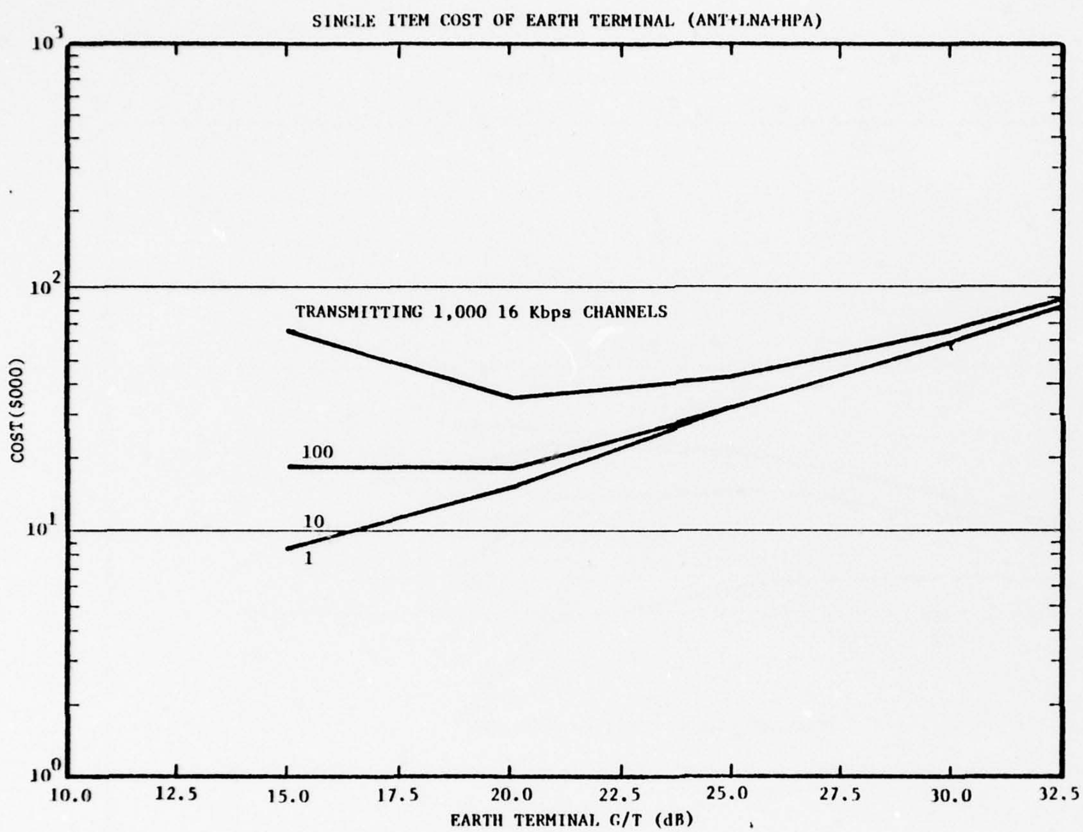

FIGURE B.2.5 FDMA X-BAND

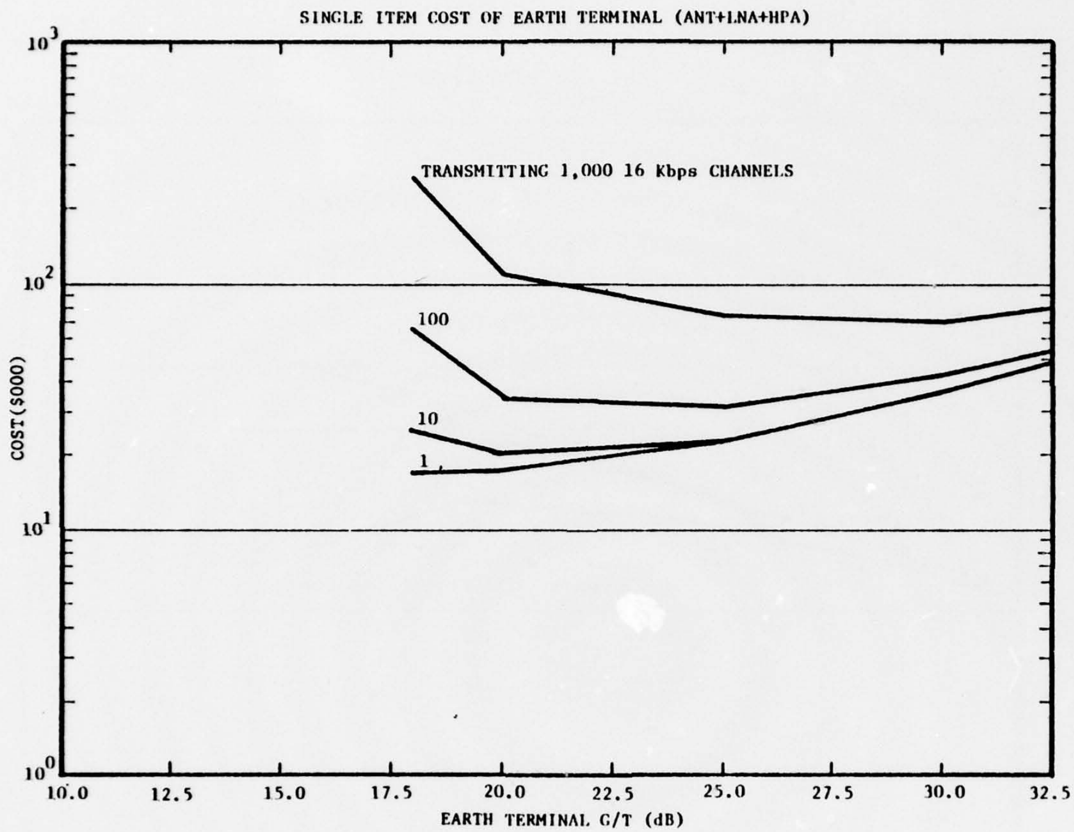

FIGURE B.2.6 FDMA K_u-BAND

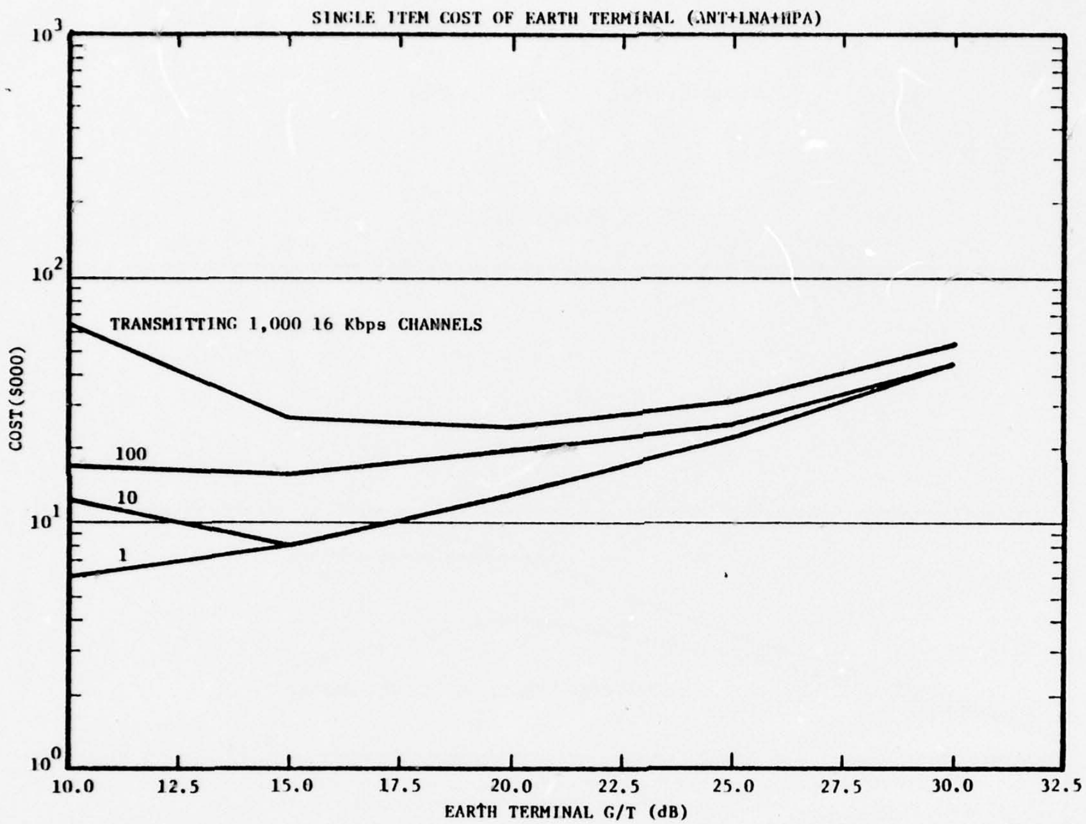

FIGURE B.2.7 TDMA C-BAND

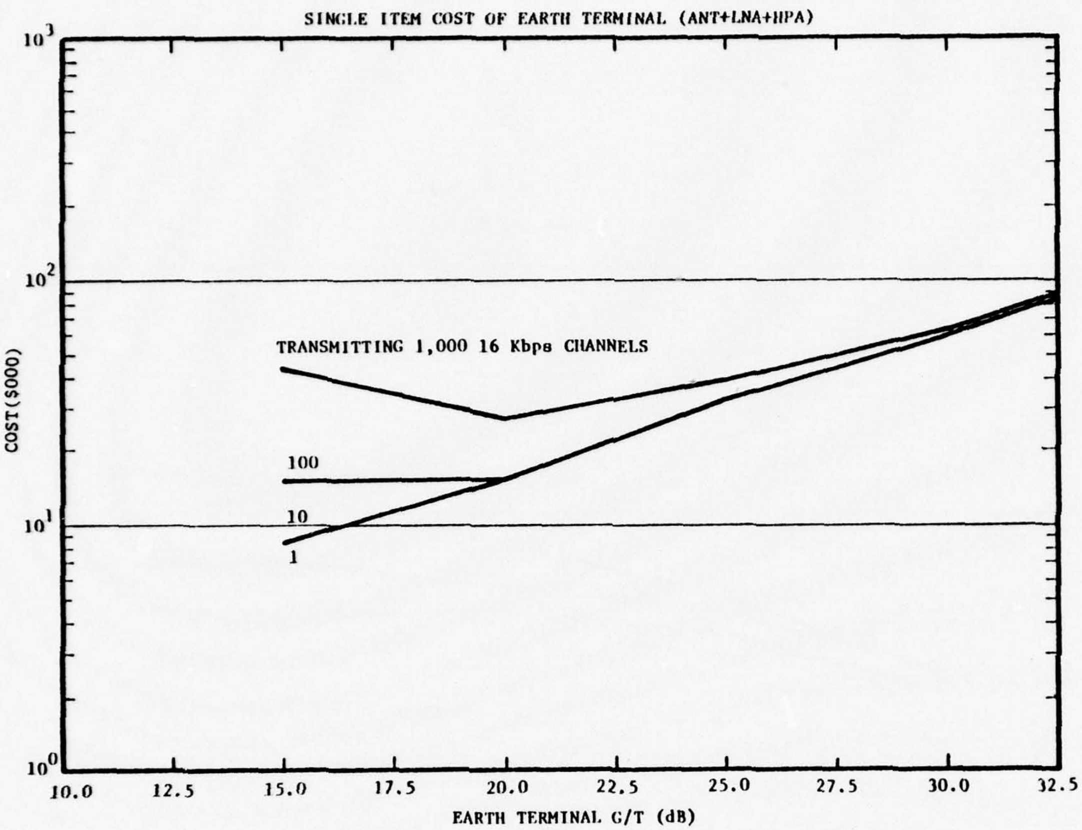

FIGURE B.2.8 TDMA X-BAND

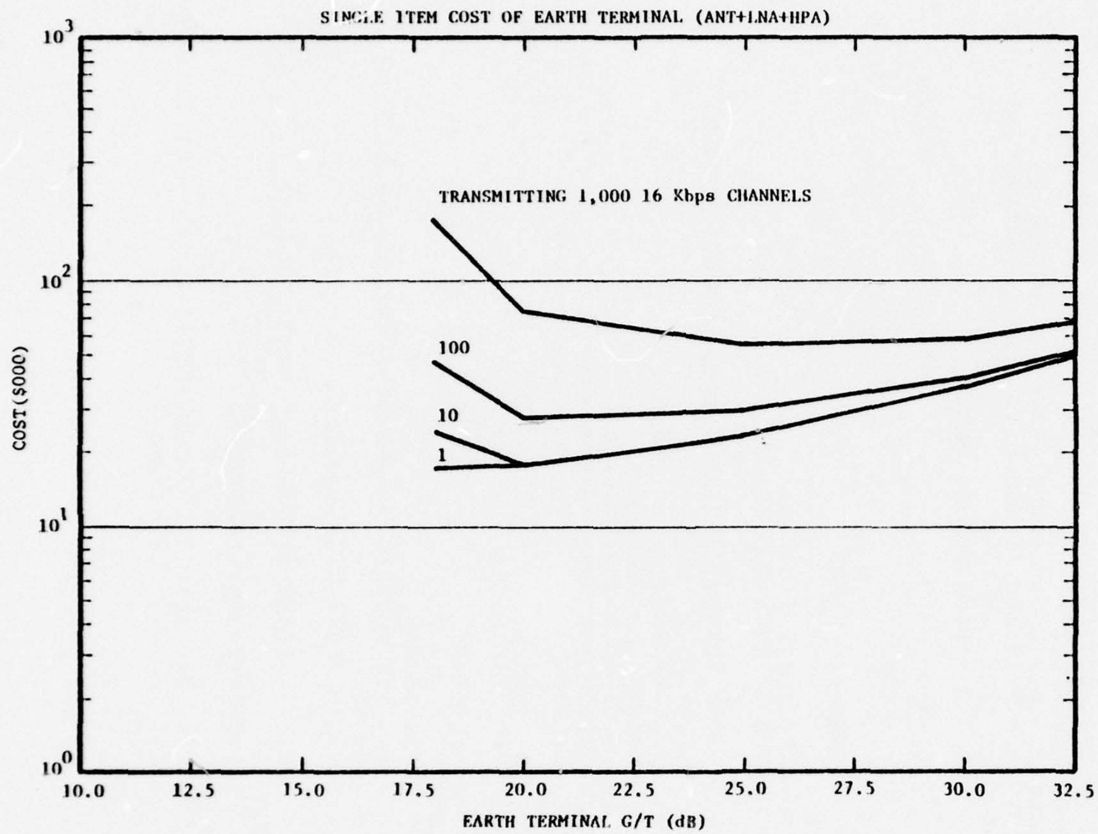

FIGURE B.2.9 TDMA K_u-BAND

```
LIST

100C... DAMA COST ANALYSIS...11-30-76 JTC
110     PARAMETER IX=3,JX=3,KX=5,LX=7
120     DIMENSION D(KX),NX(LX),PBO(IX),GTS(JX),PX(IX,JX,KX,LX)
125     DIMENSION CET(IX,JX,KX,LX),CGT(JX,KX),CHPA(IX,JX,KX,LX)
127      DIMENSION GTR(JX,KX)
130     DATA GTS/-6.,3.,-6./
140     DATA EBNU/13./
150     DATA D/5.,10.,15.,20.,25./
160     DATA NX/1,5,10,100,500,1000,8000/
170     DATA PBO/10.,3.,0./
180     DATA GTR/10.,15.,18.,15.,3*20.,3*25.,3*30.,33.,35./
190     DATA CGT/4.,6.5,15.,6.,13.,15.5,11.,30.,21.,20.,58.,35.,43.,87.,65./
200C....I=1 SCPC; 2 FDMA ; 3 TDMA
210C....J=1 C BAND ; 2 X BAND ; 3 KU BAND
300     DO 10 I=1,3
310     DO 20 J=1,3
312     PRINT:
320     PRINT:I,J
325C....K FOR ANTENNA DIAMETERS
330     DO 30 K=1,5
335     PRINT 50,GTR(J,K),CGT(J,K)
337C....L FOR NUMBER OF CHANNELS PER ET
340     DO 40 L=1,7
350     PX(I,J,K,L)=EBNU-GTS(J)-228.6+42.01+10.*ALOG10(NX(L))
360&-20.*ALOG10(D(K))+176.17+6+PBO(I)
370     PX(I,J,K,L)=10.**(PX(I,J,K,L)/10.)
420     IF(PX(I,J,K,L)-5.) 440,425,425
425 425 IF(PX(I,J,K,L)-10.) 450,430,430
430 430 IF(PX(I,J,K,L)-20.) 460,470,470
440 440 CHPA(I,J,K,L)=2.
445     GOTO 485
450 450 CHPA(I,J,K,L)=5.
455     GOTO 485
460 460 CHPA(I,J,K,L)=8.5
465     GOTO 485
470 470 IF(J-1) 475,475,480
475 475 CHPA(I,J,K,L)=45.*PX(I,J,K,L)**.15-10.*PX(I,J,K,L)**.46
476&+PX(I,J,K,L)**.855
477      CHPA(I,J,K,L)=CHPA(I,J,K,L)/4.712
478     GOTO 485
480 480 CHPA(I,J,K,L)=1.15*EXP(1.6*ALOG10(PX(I,J,K,L)))
485 485 CONTINUE
490   CET(I,J,K,L)=CHPA(I,J,K,L)+CGT(J,K)
500 40 CONTINUE
600     PRINT 55,CET(I,J,K,1),CET(I,J,K,2),CET(I,J,K,3),CET(I,J,K,4),
601&CET(I,J,K,5),CET(I,J,K,6),CET(I,J,K,7)
610 50 FORMAT(2F10.2)
615 55 FORMAT(2X,7F10.2)
630 30 CONTINUE ; 20 CONTINUE ; 10 CONTINUE
640 60 FORMAT(F16.8,F15.8)
650     STOP ; END

ready

*
```

In the derivation of these figures, assumption is made that equal cost prevails for pulsed HPA whose average power is the same as that of CW HPA. The assumption is due to the inclusiveness of vendor survey regarding to this matter, although pulse HPA are generally less costly than CW HPA at high power level (e.g., 1000 watts).

B.2.4 Common Equipment Cost and Other Cost

In addition to the cost of antenna, LNA and HPA, common equipment cost per earth terminal (excluding voice processing) are assessed at \$100,000, \$45,000 and \$50,000 for SCPC, FDMA and TDMA, respectively as shown in Table B.2.2. Note that these cost figures are rough estimates based on literature study, not on vendor survey.

Let the SIC of earth terminal be denoted by expression:

$$C_E = C_{LNA} + C_{ANT} + C_{HPA} + C_{COMEQ} \tag{B.2.3}$$

where C_{COMEQ} is the common equipment cost. The total equipment cost of a N-identical earth terminal system is then

$$C_{ET} = C_E \quad Y^{\log_2 N} \tag{B.2.4}$$

as per discussion of Section B.1.1 of this Appendix where Y is a discount factor. Additional cost of initial system activation is also considered. Using the DCA circular 600-60-1, May, 1976 as a guide, these activation related costs are tabulated in Table B.2.3.

The total N-identical terminals network cost estimate is then

$$\begin{aligned} C_T &= C_{ET} + C_A \\ &= 2.391\ C_{ET} + 2.5\ C_E + C_{ET} \\ &= (3.391 \times Y^{\log_2 N} + 2.5)\ C_E \end{aligned} \tag{B.2.5}$$

where C_E is given by Equation B.2.3. From the above equation, cost of system activation C_A appears to occupy a large portion of total cost C_T (approximately 70%). It seems reasonable to assume that this percentage will drop with the future systems having larger number of smaller ET's with more modular plug-in type components that are easily transportable. This is an area we consider important and should be further studied.

TABLE B.2.2 COMMON EQUIPMENT COST MODEL

ITEM	COST		
	SCPC	FDMA	TDMA
Orderwire Source		3,500	3,500
Data Source	3,500	3,500	
Converter	3,000	3,000	3,000
Modem	20,000	20,000	25,000
Frequency Standard		2,500	2,500
Equipment Rack & Wiring	10,000	10,000	10,000
Orderwire Processing		1,000	1,500
Other Items	1,500	1,500	3,500
SCPC Equipment	62,000		
TOTAL	$100,000	$45,000	$50,000

TABLE B.2.3 COMMON EQUIPMENT COST

ITEM	COST
Test & Support Equipment C_{1E}	0.15 C_{ET}
System Test, Evaluation & Mgmt C_{2E}	0.25 C_{ET}
Documentation C_{3E}	2.5 C_{ET}
Operational Site Activiation C_{4E}	0.07 C_{ET}
Initial Spare & Repair C_{5E}	1.8 C_{ET}
Transportation of ET	0.1 C_{ET}
Transportation of Test Equipment	0.14 C_{1E}
Total C_A	See Equation B.2.5

B.2.5 Operation and Maintenance Manpower Cost

The DCA Circular 600-60-1 stipulates that the manning level for each ET is about the equivalent of 20, representing three shifts of four-man team of operation and another two four-man teams on training and reserve. As will be shown in later sections, the cost of operation and maintenance (O&M) manning dominates the overall satellite-earth terminal annual system cost when the number of ET's are large. In view of current trend of deployment of large numbers of small ET's, unattended or minimally attended ET's should be investigated.

B.2.6 Annual Satellite/Earth Terminal System Cost Estimation

A computer program capable of evaluating the annual system cost of a satellite/earth terminal system has been written, based on the cost modeling discussed in Sections B.2.1-B.2.5, and Section B.1. The program, called DAMA 14, is written in FORTRAN IV and is executable in time shared mode. Essentially, the program computes the annual system cost as a function of the number of earth terminals and the traffic loading. A flow diagram is shown in Figure B.2.10a followed by a program listing of DAMA 14 (Figure B.2.10b).

Using DAMA 14, the annual system cost estimates of satellite/earth terminals are made with respect to the following multiple access techniques for C, X and K_u-band:

SCPC
FDMA
TDMA
ALOHA
SLOHA

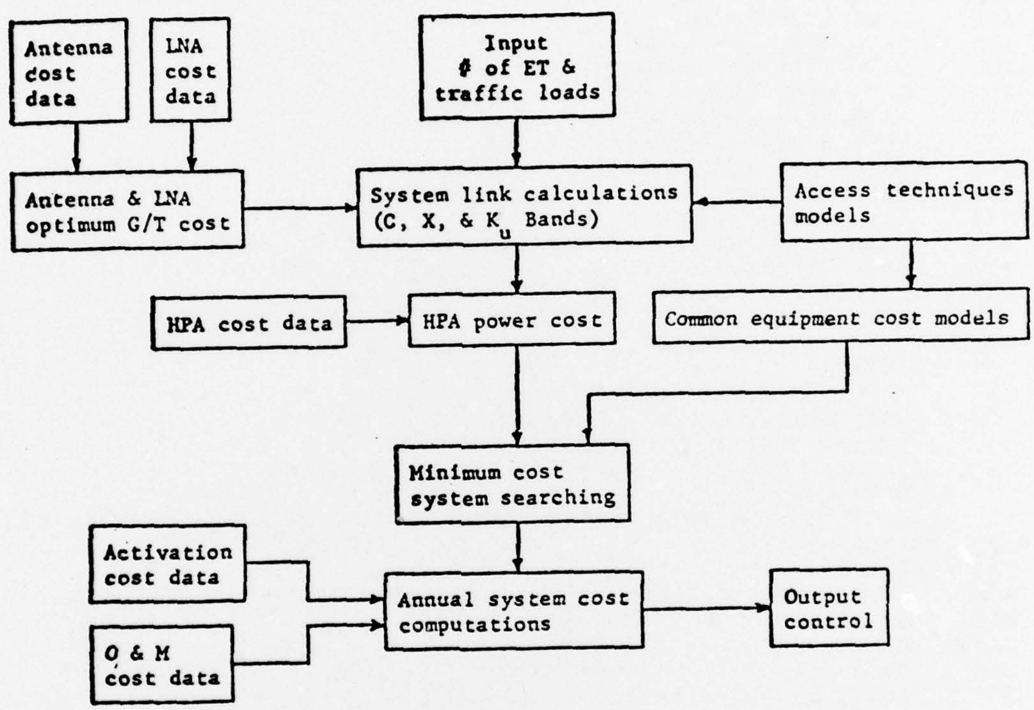

FIGURE B.2.10 FLOW CHART OF DAMA 14

```
100C... DAMA COST ANALYSIS...11-30-76 JTC
105C....DAMA14...THIS PROGRAM SEARCHES MINIMUM COST SYSTEM AS A FUNCTION
106C....OF NUMBER OF EARTH TERMINALS. C-BAND,X-BAND & KU-BAND CONSIDERED.
110      PARAMETER IX=3,JX=3,KX=6,LX=7
120      PARAMETER MX=7
130      DIMENSION D(KX),NX(LX),PBO(IX),GTS(JX),PX(IX,JX,KX,LX)
140      DIMENSION CET(IX,JX,KX,LX),CGT(JX,KX),CHPA(IX,JX,KX,LX)
150      DIMENSION GTR(JX,KX),NCH(JX,MX),NET(JX,MX,LX)
160      DIMENSION CT(JX,MX,LX),CTOL(JX,MX,LX),GTM(IX,JX,LX),CSAT(JX)
170      DIMENSION CMIN(IX,JX,LX),ADD(JX,MX)
180      DIMENSION NY(LX),CCOM(IX)
182      DIMENSION RB(JX,MX,LX),GTQ(JX,MX,LX)
186      DIMENSION RA(JX)
190      DATA GTS/-6.,3.,-6./
200      DATA CSAT/1000.,1070.,1000./
210      DATA EBNU/13./
212      DATA RA/60.,208.3,400./
220      DATA D/5.,10.,15.,20.,25.,60./
222      DATA CCOM/100.,45.,50./
232      DATA NY/1000,100,70,50,20,10,5/
240      DATA PBO/10.,3.,0./
250      DATA  GTR/10.,15.,18.,15.,3*20.,3*25.,3*30.,33.,35.,3*39./
260      DATA CGT/4.,6.5,15.,6.,13.,15.5,11.,30.,21.,20.,58.,35.,43.,87.,
262&65.,200.,500.,400./
270      DATA ADD/11.56,.56,6.05,7.56,-3.44,2.05,10.56,-0.44,5.05
280&,6.02,-4.98,-0.95,1.55,-9.45,-3.96,9.18,-1.82,3.67,4.55,
290&-6.45,.51/
295      DATA TRAFIC/15200./
300C....I=1 SCPC; 2 FDMA ; 3 TDMA
310C....J=1 C BAND ; 2 X BAND ; 3 KU BAND
320C....ANNUAL DEPRECIATION DEP=6.446 ; DEP=6.446
330      DEP=6.446
331C....TATOL>0 FOR TOTAL SYSTEM COST ; =< 0 FOR (ANT+LNA+HPA) COST
332      TATOL=2.
334      TERR=1.
336C....SYSCO=0.--NO OPERATIONAL ACTIVATION COST, YES OTHERWISE
338      SYSCO=2.
339      MANLEL=20
340      DO 5 J=1,JX ; DO 5 M=1,MX
342 5    ADD(J,M)=ADD(J,M)+2.6
344      DO 10 I=1,IX
350      DO 20 J=1,3
360      DISCT=.95
370C....K FOR ANTENNA DIAMETERS
380      DO 30 K=1,KX
390C....L FOR NUMBER OF CHANNELS PER ET
400      DO 40 L=1,LX
402      NX(L)=INT(TRAFIC*(NY(L)-1)/NY(L)**2)
410      PX(I,J,K,L)=EBNU-GTS(J)-228.6+42.01+10.*ALOG10(NX(L))
420&-20.*ALOG10(D(K))+176.17+6+PBO(I)
430      PX(I,J,K,L)=10.**(PX(I,J,K,L)/10.)
440      IF(PX(I,J,K,L)-5.) 440,425,425
450 425  IF(PX(I,J,K,L)-10.) 450,430,430
460 430  IF(PX(I,J,K,L)-20.) 460,470,470
470 440  CHPA(I,J,K,L)=2.
```

FIGURE B.2.10.b DAMA 14 PROGRAM LISTING

```
    GOTO 485
450 CHPA(I,J,K,L)=5.
    GOTO 485
460 CHPA(I,J,K,L)=8.5
    GOTO 485
470 IF(J-1) 475,475,480
475 CHPA(I,J,K,L)=45.*PX(I,J,K,L)**.15-10.*PX(I,J,K,L)**.46
&+PX(I,J,K,L)**.855
     CHPA(I,J,K,L)=CHPA(I,J,K,L)/4.712
    GOTO 485
480 CHPA(I,J,K,L)=1.15*EXP(1.6*ALOG10(PX(I,J,K,L)))
485 CONTINUE
  CET(I,J,K,L)=CHPA(I,J,K,L)+CGT(J,K)
40 CONTINUE
30 CONTINUE
   DO 35 L=1,LX ; ZZ=CET(I,J,1,L)
   DO 45 K=1,KX
   IF(CET(I,J,K,L)-ZZ) 46,46,47
46 ZZ=CET(I,J,K,L) ; GTM(I,J,L)=GTR(J,K)
47 CONTINUE
45 CONTINUE
   CMIN(I,J,L)=ZZ
35 CONTINUE
20 CONTINUE ; 10 CONTINUE
   DO 51 I=1,IX
   IF(TATOL) 52,52,53
52 CCOM(I)=0. ; TERR=0. ; 53 CONTINUE
51 CONTINUE
   DO 100 J=1,JX
   DO 110 L=1,LX
   DO 120 M=1,MX
    IF(M-1) 111,111,112
111 IY=1
   RB(J,M,L)=NX(L)*NY(L)*(.016)
   GOTO 116
112 IF(M-2) 113,113,114
113   IY=2
   RB(J,M,L)=NX(L)*NY(L)*.016+NY(L)*.01
   GOTO 116
114 IF(M-3) 117,117,118
117 IY=3
   RB(J,M,L)=NX(L)*NY(L)*.016+NY(L)*.1333
   GOTO 116
118 IY=3
   RB(J,M,L)=NX(L)*NY(L)*.016*1.3
116 CONTINUE
   GTQ(J,M,L)=10.*ALOG10(RB(J,M,L))+60.-42.04-ADD(J,M)
   SYST=1.
   KK=1
602 IF(GTQ(J,M,L)-GTR(J,KK)) 603,603,604
603 GTQ(J,M,L)=GTR(J,KK)
   LLZ=KK
   GOTO 612
604 IF(KK-KX) 611,612,612
611 KK=KK+1 ; GOTO 602 ; 612 CONTINUE
```

FIGURE B.2.10.b DAMA 14 PROGRAM LISTING (Continued)

```
      IF(GTQ(J,M,L)-GTR(J,KX)) 605,605,606
 606 SYST=0.
 605 CONTINUE
      Y=ALOG(NY(L))*1.442695
      IF(SYST) 607,849,607
 607 CT(J,M,L)=NY(L)*DISCT**Y*(CET(IY,J,LLZ,L)+CCOM(IY))
      IF(SYSCO) 701,702,701
 701 CONTINUE
      CT(J,M,L)=(.15+.25+.07+1.8+.1)*CT(J,M,L)
      CT(J,M,L)=CT(J,M,L)+(CET(IY,J,LLZ,L)+CCOM(IY))*(5.*0.5+.14)
 702 CONTINUE
      CT(J,M,L)=CT(J,M,L)/DEP
      CSAT(J)=CSAT(J)
      CTOL(J,M,L)=CT(J,M,L)+CSAT(J)*RB(J,M,L)/RA(J)
 850 CONTINUE
      GOTO 120
 849 CTOL(J,M,L)=10000000000.
 840  CONTINUE
 120 CONTINUE
 110 CONTINUE
 100 CONTINUE
      DO 901 J=1,JX
      IOUT=12+J
      DO 902 L=1,LX
      PP=CTOL(J,1,L)
      LLP=LX+1-L
      XX=FLOAT(NY(L))
      DO 903 M=1,MX
      IF(CTOL(J,M,L)-PP) 904,904,905
 904 PP=CTOL(J,M,L)
      QQ=RB(J,M,L)/RA(J)
 905 CONTINUE ; 903 CONTINUE
      QK=QQ
      CMIN(1,J,L)=PP+MANLEL*45.6675*XX
 902 PRINT 61,XX,CMIN(1,J,L),QK,NX(L)
C902 WRITE(IOUT,60) XX,CMIN(1,J,L)
 901 CONTINUE
 60 FORMAT(F16.8,F15.8)
 61 FORMAT(F8.1,E12.2,F8.2,I6)
      STOP ; END

ready

*
```

FIGURE B.2.10.b DAMA 14 PROGRAM LISTING (Continued)

For each of the 3 frequency bands, a composite minimum cost curve is constructed as a function of the number of earth terminals as shown in Figure B.2.11. As an initial estimate in this task, it is assumed in the calculations that the annual satellite charge is proportional to its required bandwidth which is assuemd to be \$1M/36 MHz for C, \$1M/125 MHz for X, and \$1M/240 MHz for K_u-band, respectively. Under these conditions, the result of Figure B.2.11 indicates a relatively moderate increase of annual system cost as the number of earth terminals N increase to about 100. But the cost escalates rapidly as N increases beyond 100. At N = 100, the annual satellite/ET system charge for C, X and K_u-band are in the order of 70 million dollars, with the system activation cost and O&M cost annualized and included. The relatively lower cost of K_u-band is probably due to the lower satellite transponder lease charge (\$1M/240 MHz) on a per MHz basis assumed. Figure B.2.11 should be updated, as needed, as more information on the component annual charges of the satellite/earth terminals system are verified in the future. By excluding the O&M cost, a plot of annual minimum system costs is presented in Figure B.2.12. Based on the cost model assumed, it is found that TDMA systems are minimum cost systems for the number of earth terminals N is less than 100. For N greater than 100, SCPC systems are minimum cost systems among the five types of access techniques considered.

Since activation cost could vary more drastically than the true equipment cost in future systems, a plot of "equipment-only" annual system cost (i.e., not including system activation cost and O&M cost) is presented in Figure B.2.13.

B.2.7 System Cost Reduction From Lower O&M Cost

In addition to previously run annual satellite/ET system cost with O&M cost at 20 man-level per earth terminal. The annual satellite earth

terminal system costs have been reevaluated at various man level for C, X and K_u-band systems:

Case 1 - 0 man-year/ET (Unattended)

Case 2 - 4 man-year/ET

Case 3 - 8 man-year/ET

Case 4 - 12 man-year/ET

The labor cost in every case run is based on DCA cost document. The resulting cost versus number of earth terminals plots are attached as shown. In addition, digital outputs for cases 1 through 4, plus the cases of 10 and 20 man-year per ET are provided. (Figures B.2.14-B.2.17).

ACQUISITION, ACTIVATION, OPERATION & MAINTENANCE)

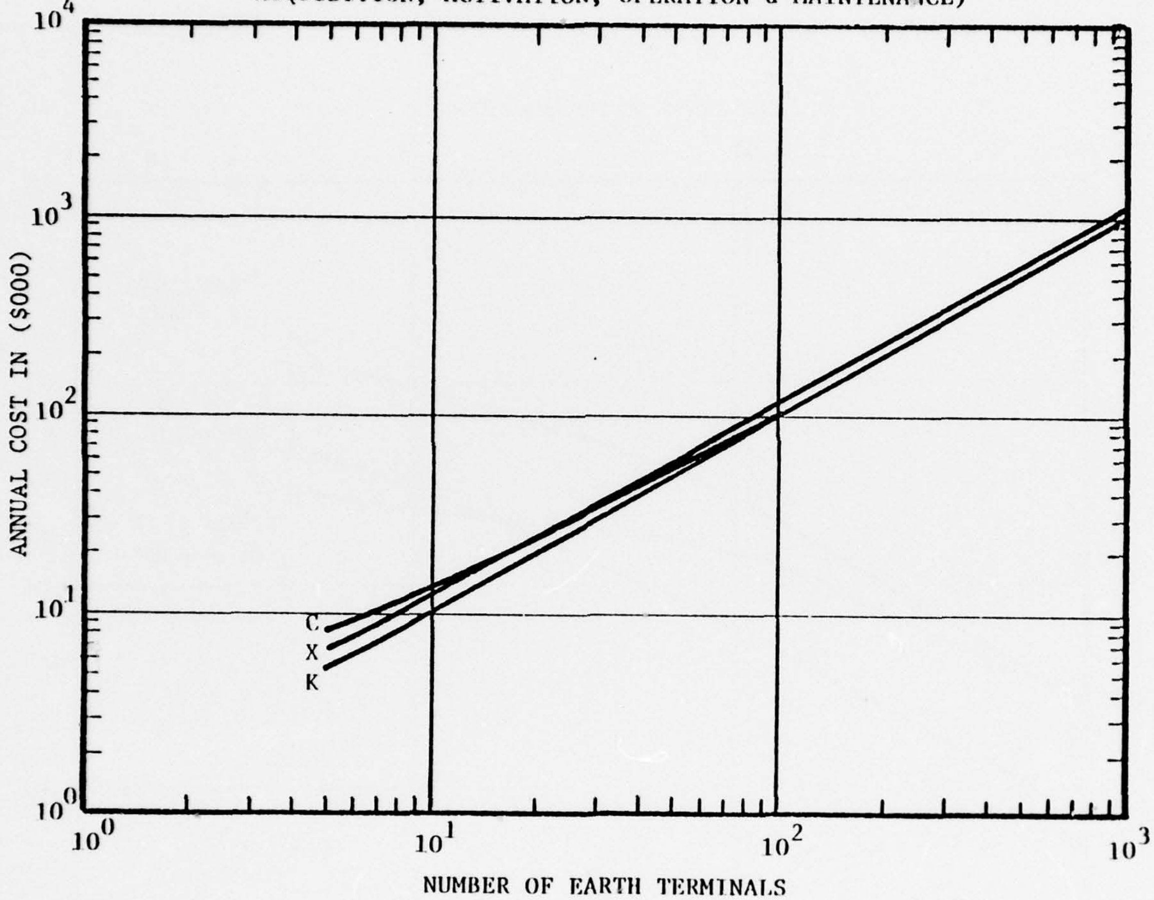

FIGURE B.2.11 ANNUAL SATELLITE/EARTH TERMINAL SYSTEM COST

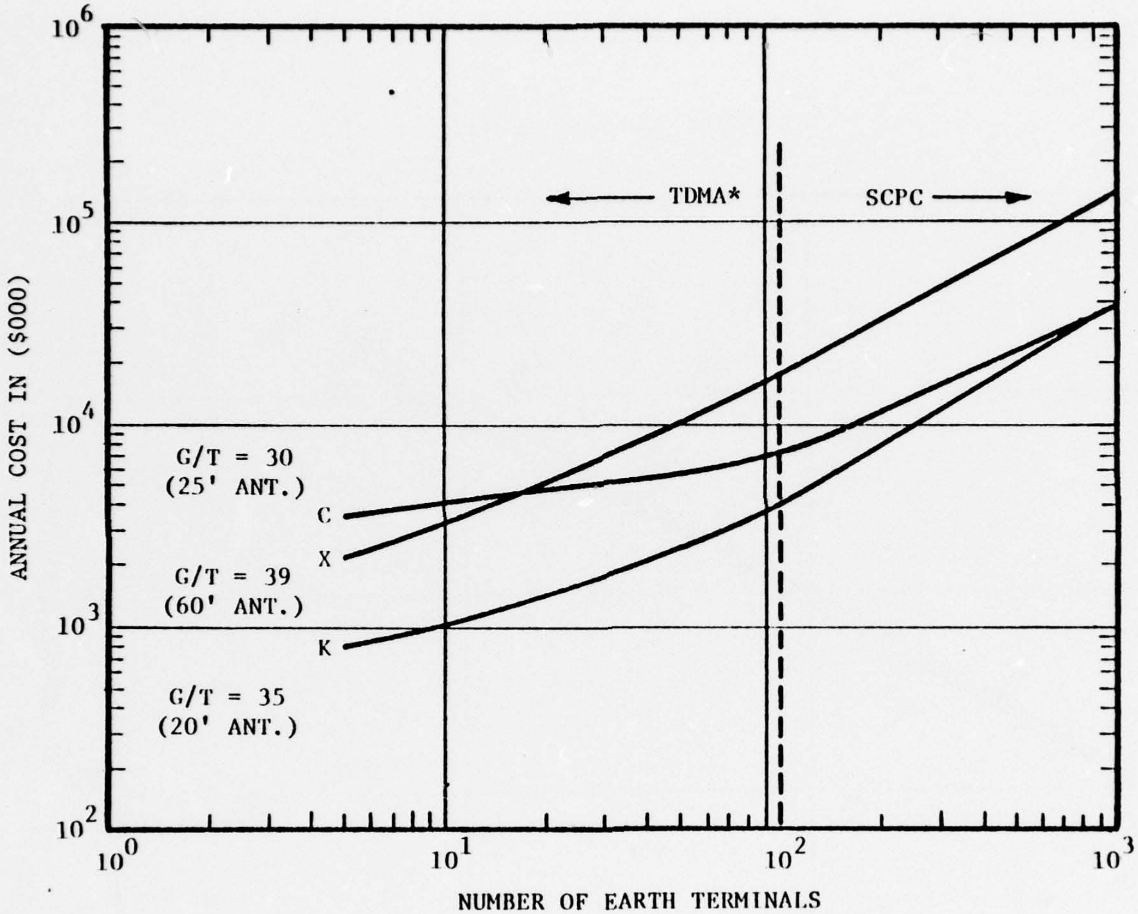

FIGURE B.2.12 ANNUAL SAT/ET MINIMUM SYSTEM COST ESTIMATE WITH OPERATIONAL ACTIVATION COST

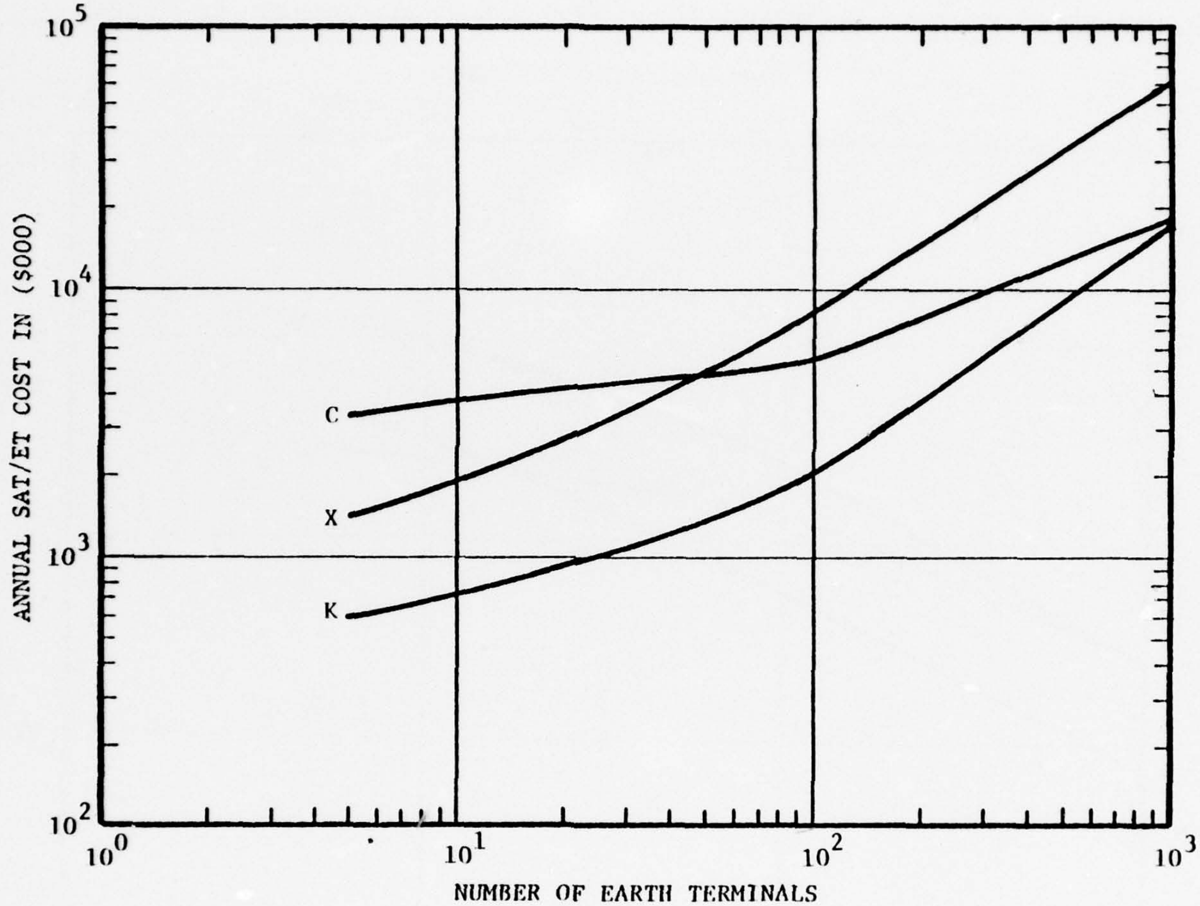

FIGURE B.2.13 ANNUAL SAT/ET MINIMUM SYSTEM COST ESTIMATE

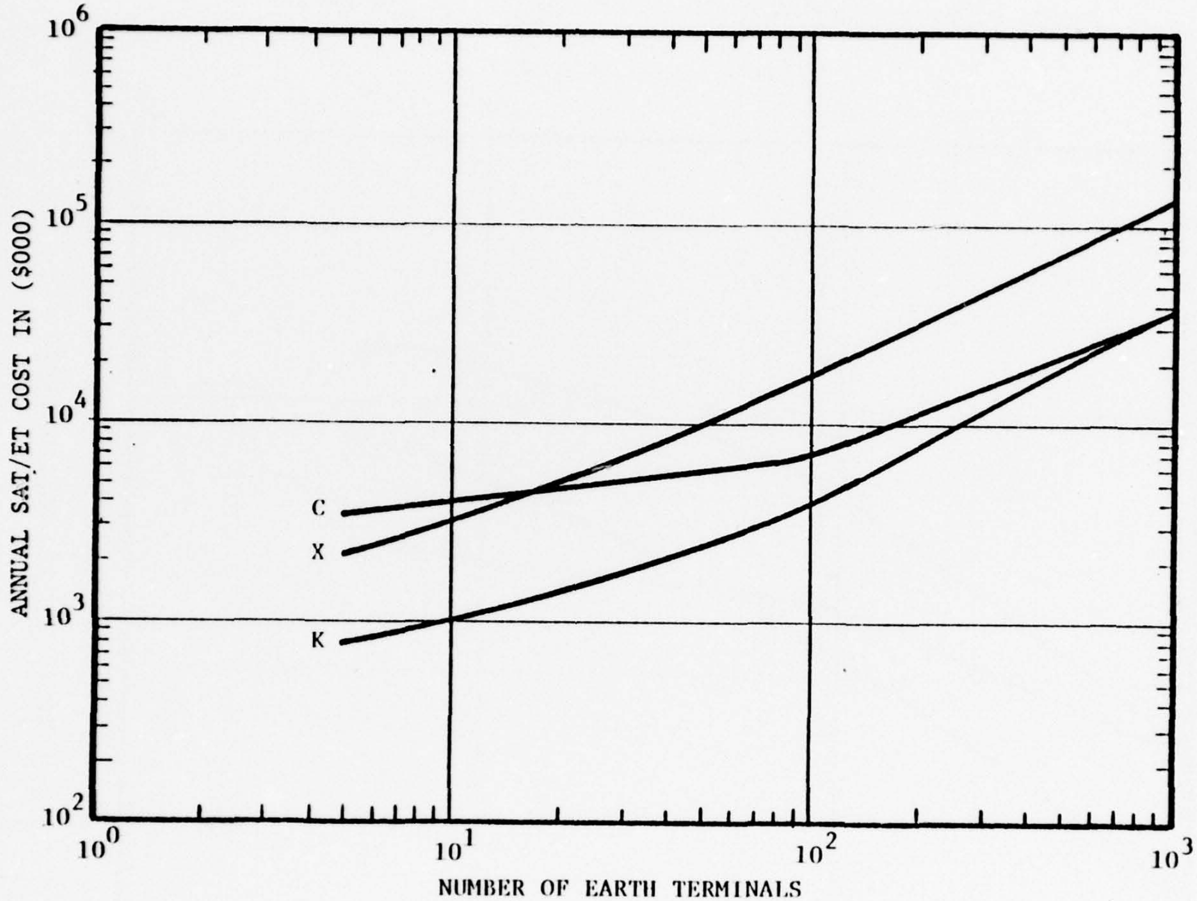

FIGURE B.2.14 ANNUAL SATELLITE/ET COST (ACO + ACTIVATION + O&M) AT 0 MAN-YEAR/ET

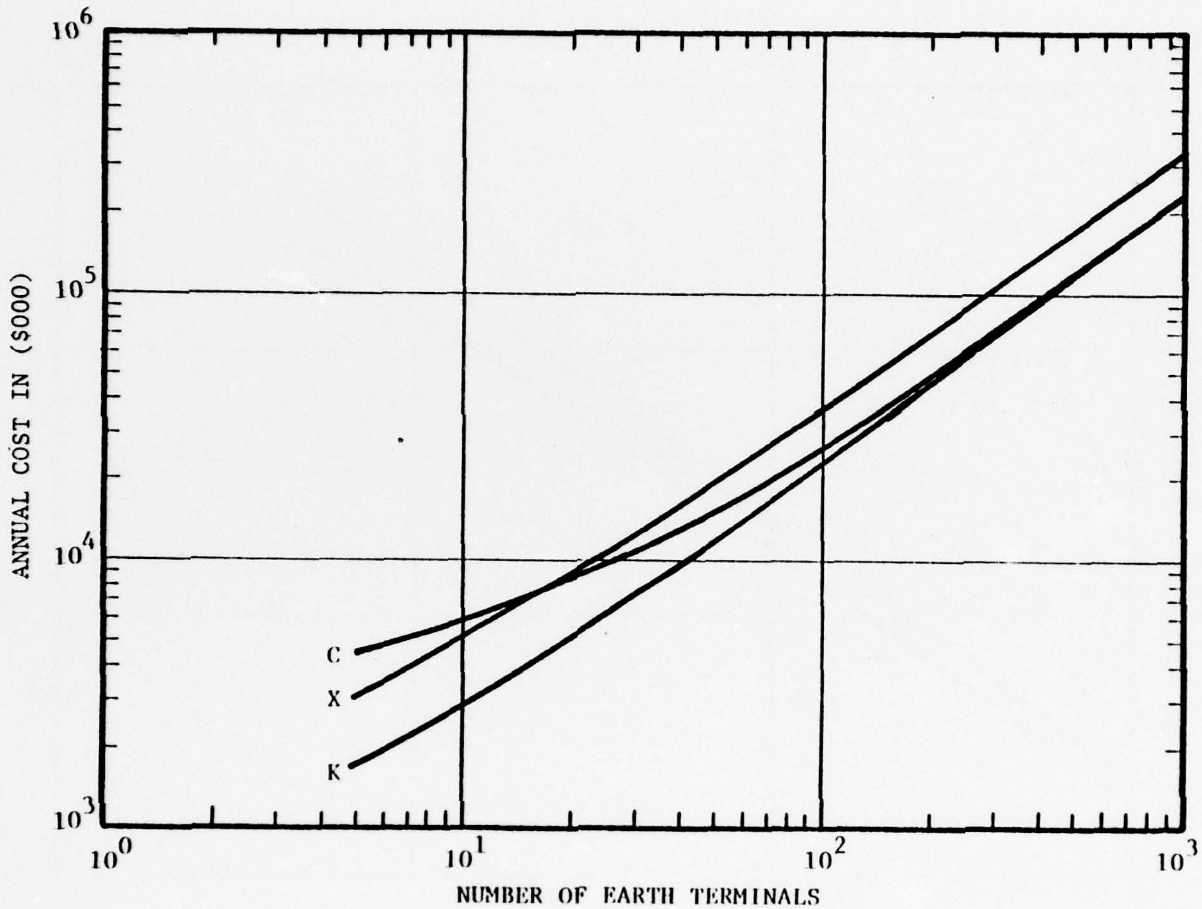

FIGURE B.2.15 ANNUAL SATELLITE/ET COST + ACTIVATION + O&M) AT 4 MAN-YEAR/ET

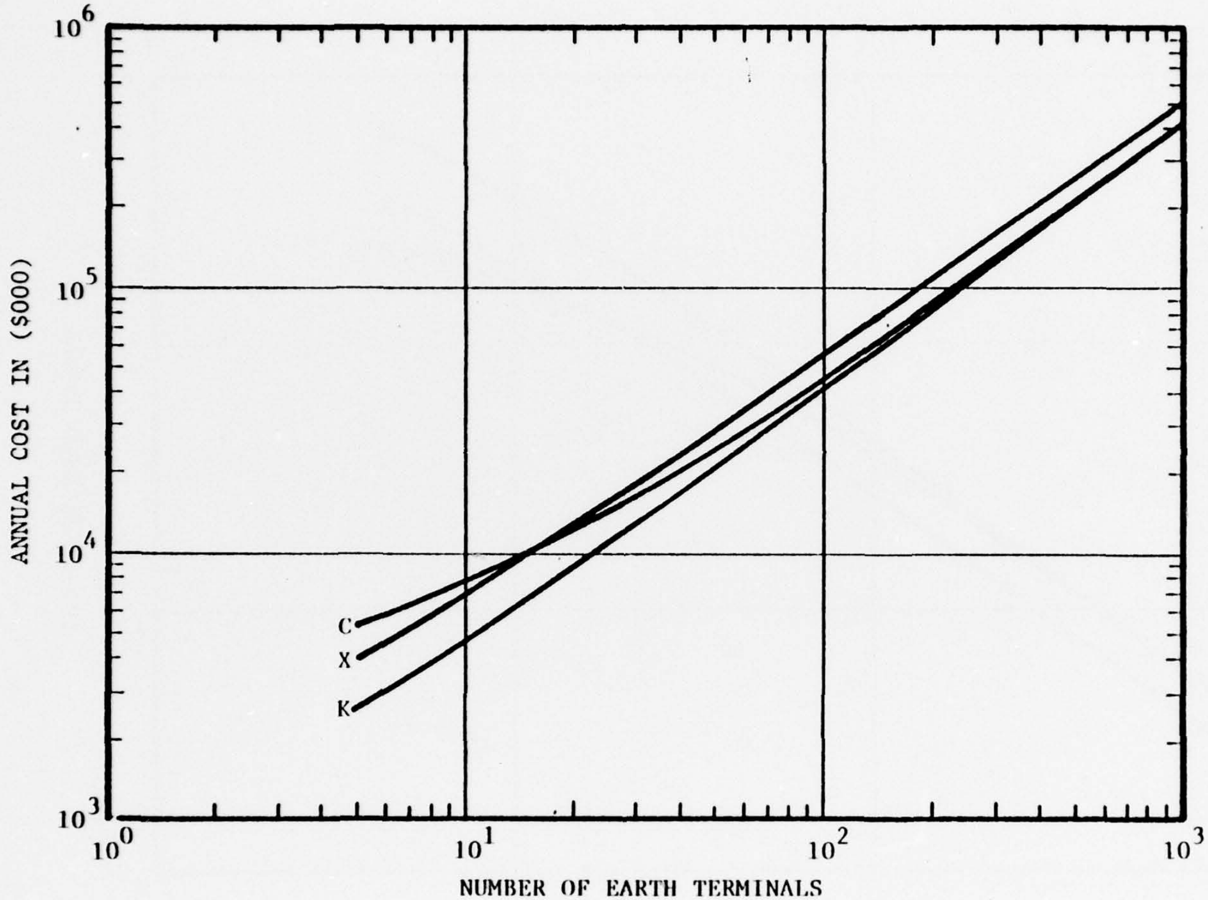

FIGURE B.2.16 ANNUAL SATELLITE/ET COST (ACO + ACTIVATION + O&M) AT 8 MAN-YEAR/ET

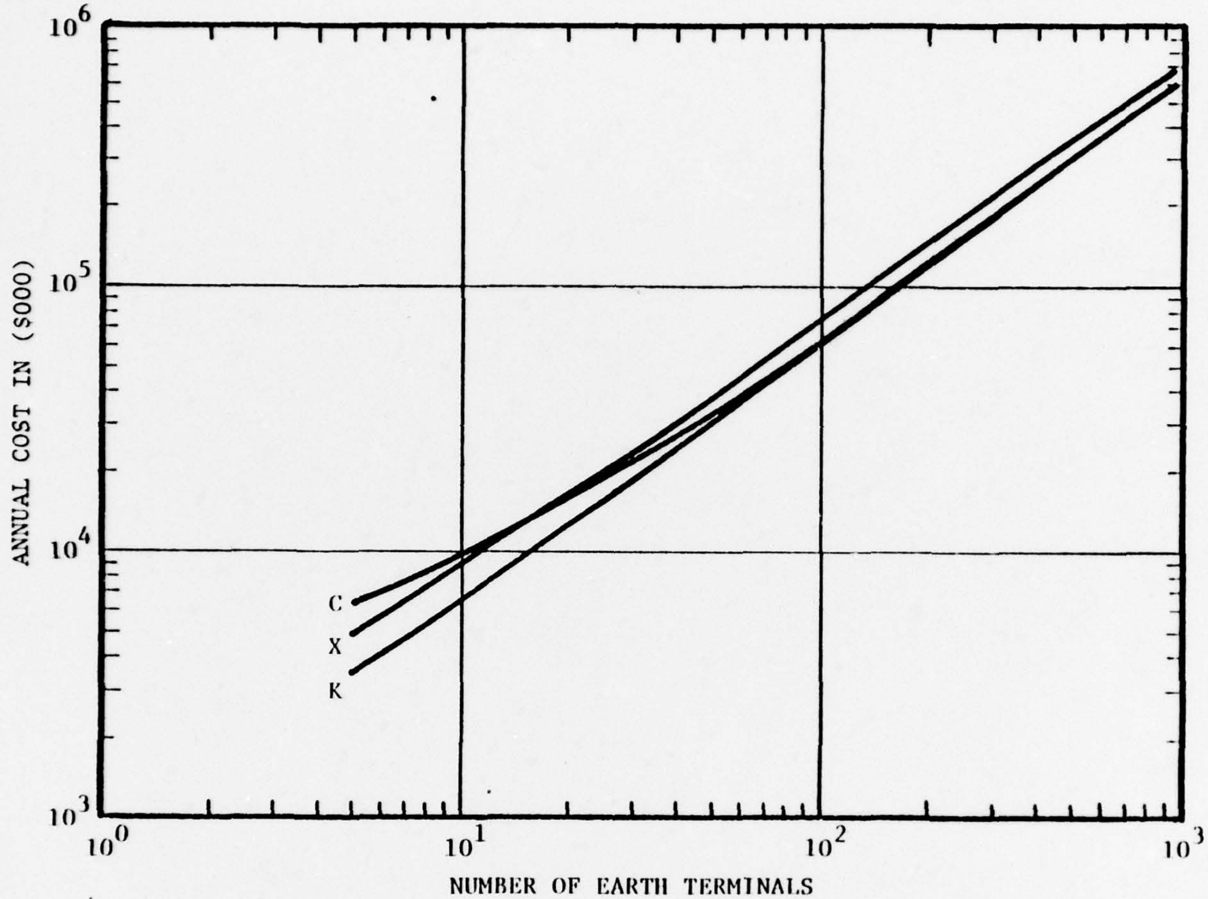

FIGURE B.2.17 ANNUAL SATELLITE/ET COST (ACO + ACTIVATION + O&M) AT 12 MAN-YEAR/ET